THE SOUND ENGINEERING MAGAZINE
JULY 1980
$1.95

db

Coming Next Month

• The August issue of **db** concludes our three-part series on Radio and Recording with special emphasis being placed on the installation (and care) of various studio systems. Also, we will present you with the best of old and new. Columnist Barry Blesser, in his second offering, tells us of his insights into digital technology in his latest chapter of *Digital Audio*. Then, Norman Crowhurst continues his detailed analysis of the specialization of human hearing in "A.G.C. and Masking." So be sure *not* to miss next month's *super*-edition of **db—The Sound Engineering Magazine.**

About The Cover

• A view of Fantasy Recording Studio as designed by acoustic consultant Tom Hidley and constructed by Sierra Audio. For more information about the construction of Fantasy Recording see this month's issue. Be sure to see August, too! Now it's time to enjoy the premiere magazine in the professional sound field—*db!*

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THE SOUND ENGINEERING MAGAZINE

JULY 1980 VOLUME 14, NUMBER 7

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David T. Gilot
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Kathy Lee
ADVERTISING PRODUCTION

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db Letters

To The Editor:

In Michael Rettinger's article "Sound Reinforcement Systems in Amphitheatres," in the May 1980 issue, the author stated that the upper sidewalls of the Tanglewood Music Shed are treated with ¾" painted fiberboard. This may have beentrue at one time, but any such fiberboard was removed before the large-scale Saarinen-Beranek and Newman renovation in 1959 and 1960. The interior finish is now all wood, except for structural steel and cables, and there is no sound-absorbing treatment of any kind present. New seating installed several years ago reduced the interior seating capacity from 6,000 to approximately 5,000, and the new seats are more comfortable. As many as 10,000 people have sat on the lawn for specific concerts.

While suspended sound-reflecting panels have been blamed for faulty acoustics in other halls, their presence in the Tanglewood Music Shed indicates that they are certainly acceptable and useful in at least some situations. A paper, "Orchestra Enclosure and Canopy for the Tanglewood Music Shed," by F. R. Johnson, L. L. Beranek, R. B. Newman, R. H. Bolt, and D. L. Klepper, in *The Journal of the Acoustical Society of America*, Vol. 33, No. 4, p. 475-481, April, 1961, describes this concert hall and its acoustics in detail.

David L. Klepper
Klepper, Marshall & King
96 Haarlem Avenue
White Plains, NY

db Replies:

The reference quoted in the Klepper letter is dated April 1, 1961.

I described the Tanglewood Music Shed from "Music, Acoustics & Architecture" by Leo L. Beranek and published by John Wiley & Sons Inc. The foreword in that book by Leo L. Beranek is dated July 1962. The ¾" painted fiberboard description for the sidewalls appears on page 145 of the book.

Possibly Mr. Klepper obtained his information after 1962 by personal inspection of the concert hall.

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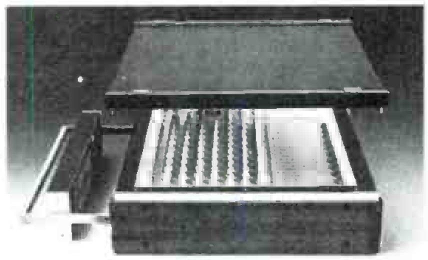

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db Calendar

JULY

25-27 **American Radio Relay League 26th Convention,** Seattle, Washington. Registration and program information may be obtained by writing: 1980 ARRL National Convention Committee, PO Box 58534, Seattle, Washington 98168.

AUGUST

1-8 **New England Conservatory of Music Summer Sessions** featuring workshops for interested musicians. Contact: New England Conservatory, 290 Huntington Ave., Boston, Mass. 02115. (617) 262-1120.

SEPTEMBER

2-4 **Prosound International Exhibition** will be held at the West Centre Hotel, London. Contact: Batiste Promotions & Exhibitions, Pembroke House, Campsbourne Road, London.

20-24 **International Broadcasting Convention** to be held at the Metropole Exhibition Center, Brighton, England. Contact: British Information Services, 845 Third Ave., New York 10021. (212) 752-8400.

24-25 **REPCON '80 Exhibit** will be held at the LaGuardia Sheraton Inn, Elmhurst, New York. For more information contact: Electronics Representatives Association, 8243 Jericho Tpke., Woodbury, New York 11797. (516) 692-5044.

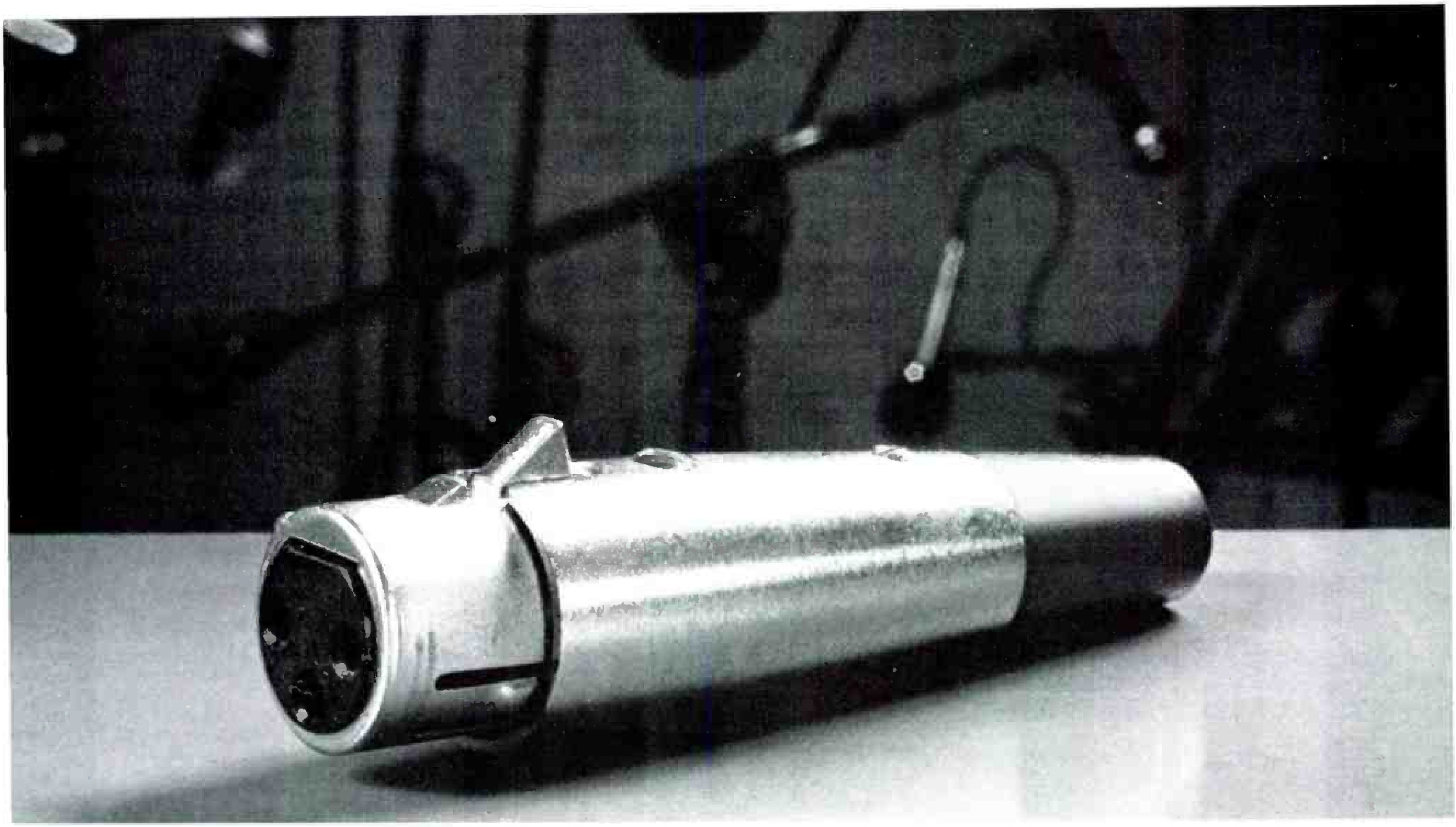
25-26 **University of Wisconsin** will hold a Recording Classical Music Workshop. Contact: Burton Spangler, University of Wisconsin, Eau Claire, Wisconsin 54701. (715) 836-2651.

OCTOBER

5-8 **National Radio Broadcasters Association (NRBA) Convention** to be held at the Bonaventure Hotel, Los Angeles. For further information contact: NRBA, 1705 DeSales (NW), Suite 500, Washington, D.C., (202) 466-2030.

14-16 **Internepcon UK '80** will be held at the Metropole Exhibition Center, Brighton, England. Contact: British Information Services, 845 Third Ave., New York, New York 10022. (212) 752-8400.

28-30 **TESTMEX Exhibition** to be held Wembley Conference Center, London. Contact: British Information Services, 845 Third Ave., New York 10022. (212) 752-8400.



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NORMAN H. CROWHURST

db Theory & Practice

A.G.C. And Other Things

• Pursuing our discussion into what may generally be termed "signal processing" one of the early forms was automatic gain control, abbreviated to AGC. At first it was applied to incoming radio signals, to compensate for variations in signal level, so the audio part of the receiver would always receive the same level of demodulated signal. Later it was used to improve the quality of audio, in the particular sense of removing unwanted "noise." It is still used that way, in a much more refined way, in Dolby systems.

What we want to look at here, is how that parallels what happens in human hearing and, for that reason, works with the human hearing faculty in improving the illusion. But first we want to go back over some experiences in early efforts in this direction, to show what we can learn from them. We have covered the subject of noise reduction in this column before, from the purely objective viewpoint. Here we are getting into the subjective, in the sense that we want to examine perceptions achieved and what we can learn from them.

EARLY AM

In the early days, of AM radio, with what today we would call low wattage stations, and of phonograph recordings on the old shellac discs, that contained an abrasive filler, whose purpose was to keep the needle (the forerunner of today's stylus) sharp, although many users suspected it was to wear needles out, all forms of reproduced audio had what today we would call extremely limited dynamic range: there was not much "headroom" between the noise floor and the maximum power or modulation ceiling.

Because of this, popular music, such as dance bands, was more popular than classical music, such as orchestras. Dance music is played at a relatively constant level, and this could be adjusted to be relatively close to the "ceiling," so the listener was not conscious of the "floor." But orchestral music could vary between a crescendo involving all the instruments in the orchestra, down to a single instrument playing solo and pianissimo, a range of 70 db or more.

"GAIN RIDING"

With those early systems, no way could orchestral music be accommodated without at least a certain amount of "gain riding." If the gain was not changed, and the solo pianissimo passages were played at the same setting used for the crescendo passages, the solo parts would be completely lost, inaudible, down in the noise: the listener would have no way of knowing they were even there.

So, to produce recordings or transmissions of orchestral music, the audio man had to "gain ride," pulling the gain down on crescendo passages, and bringing it up, quite a long way, on pianissimo passages. That was the only way the receiver, or the reproducer, could even hear the program. It was a form of human "compression." It compressed what was really a dynamic range in the order of 70 dB, down to a range of perhaps less than 40 dB.

Skillfully performed, the listener could not detect that it was being done, but at the same time, it did not reproduce the realism of the original performance although the listener might have difficulty saying why. Then came electronic compression and expansion.

FORERUNNER TO DOLBY

The compression sought to automate what the audio gainrider had done manually. And by careful design it could

do a fairly good job. The next step was to design an expander that did precisely the opposite: ran the automatic gain riding in the reverse direction, so as to restore the original dynamic range. What we want to comment on here is the illusion that such a complete system created.

In many respects, it was like Dolby is today. It just had some weaknesses in it that Dolby was invented to overcome. But with regard to dynamic range, ignoring such things as harmonic or intermodulation distortion, it did the same thing as Dolby now does. And the first time you heard it, after being accustomed to the limitations of the system in which you were constantly either bumping your head on the ceiling, or losing your contact lenses on the floor, it was like a miracle: the space that formerly was less than 40 dB "high," now seemed to accommodate 70 dB in comfort.

If you ran a direct comparison, it was as if the noise disappeared, by a miracle. True it wasn't perfect, which was why Dolby had to be invented later. But it showed what could be done. Now, how does that tie in with what happens in the human hearing faculty?

THE HUMAN AGC SYSTEM

What compression and expansion, or the Dolby system at a later stage, provides, is a form of automatic gain control, A.G.C. Does our human hearing systems have an A.G.C. system? What controls our perceived sense of loudness? Fletcher and Munson did some early work, making subjective measurements to determine what our perceived sense of loudness is, for the average person, but that does not explain it. Other researchers since Fletcher and Munson have virtually confirmed their results, with only very minor differences.

It is pretty apparent, from all the research done, that our hearing faculty performs as a frequency content analyzer with some very sophisticated capabilities. It can recognize various sources of sound, such as different types of musical instruments, or different human individuals (not only male or female, but down to the identity of a person you know), by referring this analysis to data in our memory bank. At the same time, it uses the same content to identify a tune, or the words that person says.

But the capability we are particularly considering here, is its ability to make adjustments for the level of the acoustic vibrations received, thus to be analyzed. The level of acoustic vibration that begins to cause us distress is at one trillion times the energy level of an acoustic vibration that is just audible, on average. In terms we use every day, as audio people, it is 120 dB.

The mind boggles at what our hearing faculty does as a matter of course, every day. And when you come down to it, it is this enormous capability of our hearing faculty that led to using compression and expansion and, later, the Dolby system, to enable electronic reproduction systems to even come close to simulating the dynamic range that we listen to every day, without a second thought. How does it do this?

It has been fairly reliably established that the basilar membrane, which traverses the spiral length of the cochlea, separating the vestibular scala and the tympanic scala, consists of transverse fibers, each of which is "tuned" to a different frequency and thus will resonate, in conjunction with the fluid in the vestibular and tympanic scala, to its own frequency. Such vibration is detected, in turn by the hair cells contiguous to that particular fiber, which initiate nerve impulses to be sent to the brain.

FUNCTIONS OF THE MIDDLE EAR

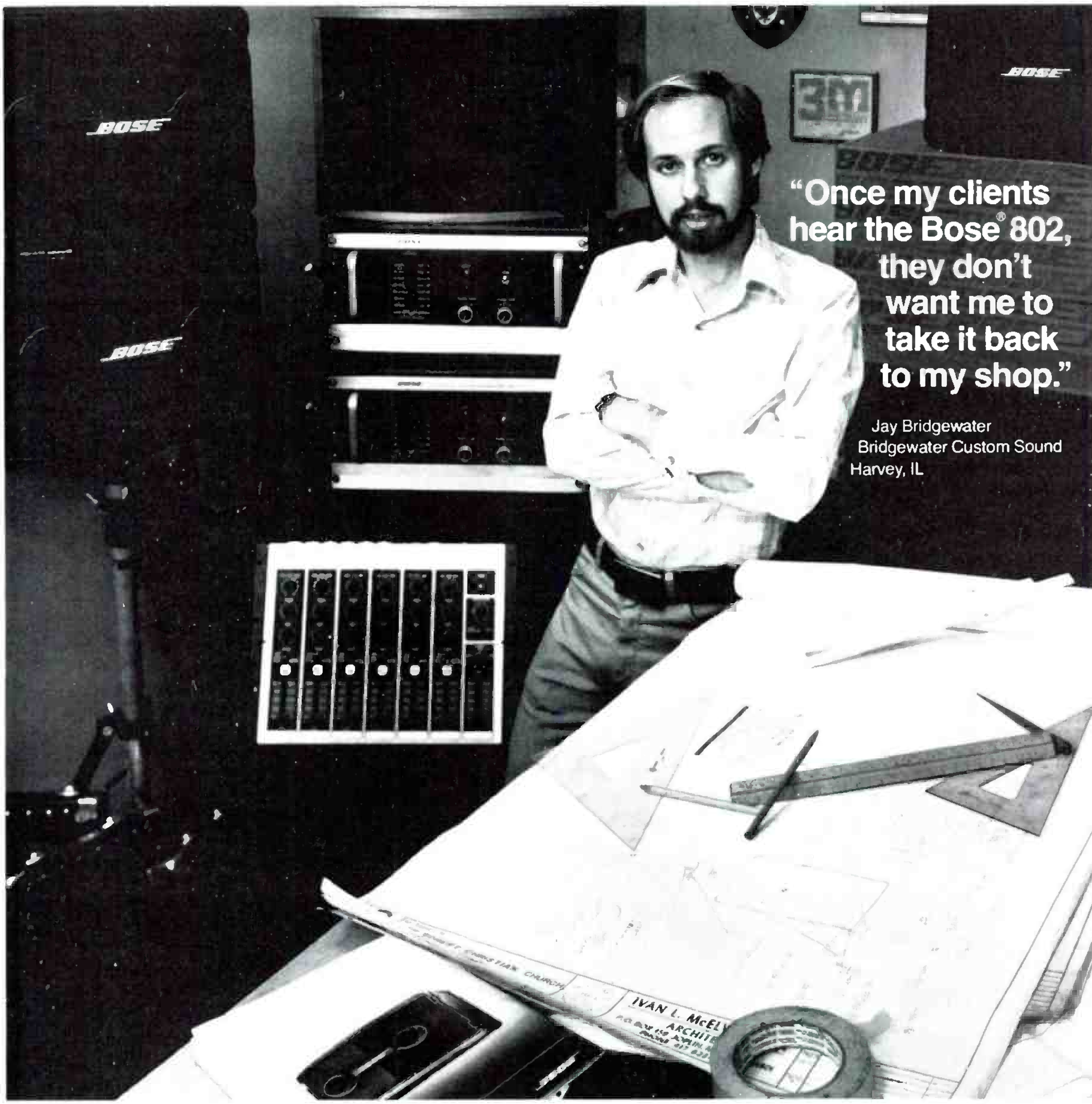
So far so good, and that would about completely explain it, *if* somewhere in the system there was an AGC control, to ensure that vibrations to be analyzed were kept within a range of 10 to 20 dB. No way could the system, if that was all there is to it, handle a range of 120 dB. So where is the AGC?

It seems that part of it is in the middle ear, where the three little bones are, that couple the eardrum to the membrane across the oval window, which communicates the vibration to the fluid in the vestibular scala. The obvious function of this little linkage of bones, is to serve as a wide-range acoustic transformer, to match acoustic vibrations in air, that are in contact with the eardrum, to acoustic vibrations transmitted to the fluid in the vestibular scala, which is a great many times denser than air.

But there are two muscles associated with these bones, which serve to do more than merely support the bones in their proper position to do the transformer job. It appears that they either behave as a variable ratio control, or else they impose variable loading on the acoustic vibration, before delivering it to the inner ear. Possibly what they do is really some of both. In any event they are the first step in the AGC process, so that the fluid in the inner air does not have to accommodate the enormous range of acoustic pressure variations to which the eardrum is subjected.

THE BONE STRUCTURE AS A TRANSFORMER

If you think about it, it is fairly obvious that the eardrum must be most sensitive to acoustic vibrations, when you are listening to sounds close to the threshhold of hearing. And if it retained that sensitivity up to sound pressures a million times as great (120 dB), the eardrum would surely rupture due to



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excessive amplitude of vibration. It is prevented from doing this by increased "loading" on the eardrum, communicated to it by the bony acoustic transformer.

At the same time, the amplitude of movement transmitted to the oval window is reduced, relatively. Suppose the increase in loading on the eardrum, due to a level increase of 120 dB, is of the order of 60 dB: then the increase in amplitude of movement of the eardrum would rise by only 60 dB, instead of 120 dB. And perhaps the amplitude range of movement communicated to the oval window could be reduced to 30 dB, by a 30 dB change in transformation ratio.

If something like that happens, part of our impression of loudness would be due to the sensing mechanism that detects the need and controls the muscles that effect this change. This is just like the way our eyes detect brilliance of light. Both of these functions take a little time, due to some kind of reaction effect, while the muscles effect the adjustment.

If you are suddenly subjected to a much louder noise, it produces a greater effect than if it is gradually increased in intensity. And if you suddenly go from low intensity illumination, to lighting several orders of magnitude greater, or vice versa, your sight takes time to adjust. For some people this may be seconds, for others, minutes. This is a typical reaction time, characteristic for each individual, of that particular function.

Obviously, if loading the eardrum decreases its sensitivity as higher sound pressures are encountered, this cannot be frequency selective: an increase in sound level at one frequency will reduce the sensitivity of the ear to sounds at all frequencies. This is the first cause of the masking effect, first measured by Fletcher and Munson.

THE MASKING EFFECT

But measurements of masking effect show that the amount of reduction in sensitivity at other frequencies, also depends to some extent on the proximity of the frequency being masked (the one of lower intensity) to the one doing the masking (at higher intensity). This would not be explained by changing of the loading at the eardrum.

A closer examination of the basilar membrane finds it to have a highly complicated set of mechanism associated with the hair cells used to detect its resonant movement. Incidentally, a closer examination also reveals that the length of the transverse fibers, free to vibrate, is greater at the apex of the helix, than near the base, which conforms with what we know about resonant systems. The shorter fibers, close to the windows, and requiring only a very short column of the fluid to vibrate with them, respond to the extremely high frequencies and the longer fibers, near the apex, and requiring the full column length of fluid above and below to vibrate with them, respond to the lowest audible frequencies.

The ear is most critical through the mid-range, neither at the highest nor the lowest audible frequencies which, if you think about it, is logical. But how does sensitivity change, or impression of loudness change, in this part of the ear, to account for the effects observed in measurements? If the tension on the basilar membrane were increased, to reduce its movement, that would also change its resonant frequency to a marked degree.

It has been observed that the pitch identified by a certain frequency does change a little with the intensity of the vibration. But this change is quite small, which it would not be, if the tension of the responding mechanism were changed. So apparently the main effect of the control mechanism, again, is to adjust the load, which has an effect analogous to changing the viscosity of the damping. Space having called a halt, we must pursue this another time. ■

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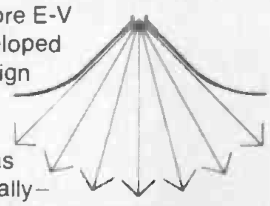

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[1] U.S. Patent Number 4071112

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N I WEINSTOCK

Videotape: Advantages and Disadvantages

INTRODUCTION

For the foreseeable future this column is going to head off in the direction of video and film, particularly video, as the visual technology of today and the future. Future columns will cover equipment peculiarities and recording technique, but a good place to start is probably with the tape itself.

VIDEOTAPE

Videotape is of course basically the same stuff as audiotape. You could take the proper width videotape, spool it onto an audiotape reel, and record on it—with superior results, one might add, unless particle orientation has been optimized for helical-scan video heads. Why superior? First because the frequency response of videotape starts about where audio's does: at the 20 or 30 Hz threshold of hearing. From there, it goes on up to 5 MHz, or two hundred times that of audiotape. Much-smaller metal particles, of a formulation very similar to ferrichrome, have to be laid on the tape to achieve this, and the tape coating must have a greater density than audiotape. The coating as a whole, however, does not need to be thicker than audiotape, nor does the base. Mr. Tome Tanaguchi, engineering director at Maxell, told me that their open-reel videotape is 20 micrometers thick, of which 15 is backing and 5 coating. By comparison, a Maxell 60-minute video cassette tape has 12 micrometers of backing and 6 of coating. And, much of the videotape in the now-professional ¾ inch U-matic format is also thinner than 1.5 mil audiotape.

EXTENDED DURABILITY

However, videotape requires a lot more durability than audiotape. In the transverse scanning formats, it moves much faster: in hellical-scan systems, there is more inherent head wear. In any format, video heads actually compress the tape. Therefore, the binding system must be much better than that needed in audio. If it is not, or if a tape has simply come in for an inordinate amount of wear, the wear can often be heard before it is seen! This is because the audio portion of all videotape is a strip just on the edge of the tape. Particularly in helical scan, the edge of the tape takes a lot of the beating.

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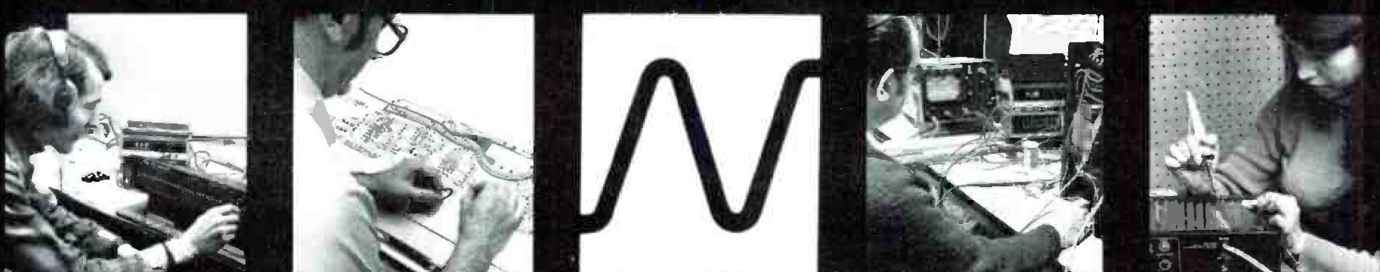

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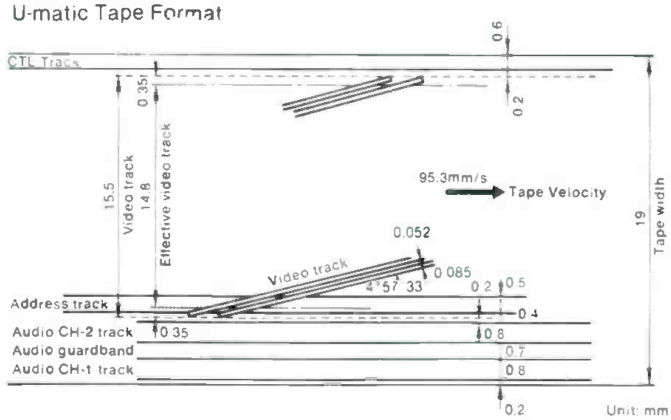
FRICTION AND HEADWEAR

"Some people think that if a videotape causes a deck to show more headwear, then it is better," says Tanaguchi, meaning that the tape is causing more friction with a machine's heads, and therefore allowing more (better) contact as they press against it. "This is very wrong information, and bad reasoning. If the heads show less wear, it's definitely better. Less friction is good for both the durability of the tapes and the heads. A well-designed binder will insure that, and we have found that the recorded image is not worse when the tape has less friction."

VIDEO LEVEL/VIDEO NOISE

Which gets us to the subject of video level and video noise. How are they different from the same terms in audio? Dispersion of magnetic particles and tape surface roughness cause a degree of distortion, and all other sources of modulation noise affect video just as audio. Then there is AC bias, or hiss, which is synonymous in video and audio. But DC noise affects video much more than audiotape. Another important difference is that audio output can be increased, in order to decrease S/N. For video, a fairly-low output level is standard, and the only way to reduce the noise is to improve the tape and electronics.

U-matic Tape Format



One of the many videotape formats is the U-matic. U-matic as depicted above is picking up popularity in many applications.

"HEARING" NOISE WITH YOUR EYES

Finally, we come to the shared border of technology and psychology. It seems that, golden ears or no, people perceive noise more easily with their eyes than their ears, and a higher level of noise is tolerable in the audio portion than in the video portion of a program. Maybe part of this is caused by the built-in inferiority of the video display, but audio "display" is pretty lousy in the TV world, too. The phenomenon of noticing more with the eyes than the ears seems demonstrable in film also, where image quality is much greater than video. In fact, the quality of even a 16 mm film image remains a more-precise rendering of reality than standard American television's: something we will discuss a bit more later on, as the potentials of film and video will be compared. But for now, a lot more needs

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They offer the control, reliability and durability that are as professional as you are.

The active crossover networks: F1040 & F1030 These frequency dividing networks offer the superior sound and control of bi-, tri-, and quad-amplification. They also offer better specs, better frequency response, and more headroom than lower priced competitive models.

The analog delays: E1010 & E1005 The creative applications of these two analog delays are almost endless. They offer echo, flanging, reverb, time delay, and double-tracking—just to name a few. And being analog, these delays retain the original audio signal for a true musical sound.

The graphic equalizer: Q1027 The Q1027 monaural ⅓ octave EQ provides virtually infinite tonal control, from subtle to dramatic. A center detent position on each filter control removes that filter from the signal path, eliminating unnecessary phase shift. The Q1027 offers many attractive features, not the least of which is its reasonable price. It even includes rack-mount and acrylic security cover.

All Yamaha signal processors are designed to give you total command over your sound system with accurate, repeatable set-ups. The quality components, quality control and rugged construction assure you years of trouble-free operation—either on the road or in fixed installations.

For complete information, write: P.O. Box 6600, Buena Park, CA 90622. In Canada, write: 135 Milner Ave., Scarb., Ont. M1S 3R1.

Because you're serious.

YAMAHA

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to be said about the tape, like the very confusing subject of the many formats it comes in.

VIDEO TAPE FORMATS

In a word, there have been far too many tape formats for video's own good, but now some standardization seems to be occurring. Two-inch tape remains the broadcast standard. One- and two-inch tapes are recorded upon by a few transverse scanning heads (in the case of two inch tape, four—hence, quadruplex or "quad.") The transverse scanning head embodied the first great breakthrough in video: the heads, aligned on a drum, move against the tape. By substituting rotating heads for the high transport speed necessary for high frequency recording, the bulk of tape needed was no longer too huge to manage.

The machinery of transverse recording, particularly the manner in which the resultant four recordings (one from each head) are lined up into one picture, would easily take all our space in itself. And, within the last few years more video professionals have become involved only in *helical* scan—a format originally devised merely for amateurs and closed circuits applications. Now, through the refinement of time-base correctors and processing amps, the signal produced (even on a half-inch portapak or VHS home recorder), can be made broadcastable. And, the tapes now made on ¾ inch cassettes can be made to look indistinguishable from two inch quadreplex. Why hadn't this been possible before? Basically, because the helical scan system created so much longitudinal wow and flutter that adequate compensation was thought to be impossible.

We haven't even mentioned all the formats yet. On the mostly-amateur end are Beta and VHS with VHS providing a much-better image. Portapaks now use ½ inch cassettes instead of open reel. I found from teaching experience that generally the most rugged and dependable portapak is Sony's, which is locked into the Beta format. In ¾ inch cassette, Sony's U-matic has conquered the field. Odd formats such as one inch, ¼ inch, IVC's ¾ inch, and others are growing extinct, although a Bosch ¾ inch format is alive and well in Europe, I'm told. It is quite easy to feed bulk tape into a U-matic cassette, if you wish to economize, but even purchased whole, the software involved isn't expensive. The hardware is another story, but the success of the cassette format, as well as smaller high-quality cameras, and the fine computerized editing consoles that are now creeping into audio, have all brought broadcast quality to many a production house, small sound studio, ad agency, and many large corporations for in-house training films, sales and public relations. The success of ¾ inch has spurred an ever larger demand for the video recording engineer, too. It has not yet, however, spurred demand for higher-quality audio in ¾ inch video. At present the width of the audio track in two inch quad is 0.07 inches—in ¾ U-matic, it's even less: in other words, not good enough for much more than a tin-eared television.

...STILL ROOM FOR IMPROVEMENT

The possibilities for audio in video are necessarily limited by the medium, and there's not much that can be done about it at this point. Videotape remains a medium of compromise, offering not nearly the visual performance of film, and not nearly the aural performance of audio tape. Image quality is going to improve dramatically with some technology just around the bend, including the disc, but all that is holding back the audio portion is the low perception of audio among those who've developed TV and video all along. Stereo videodiscs for the home should bring in high quality audio in videotape as well. And finally, computerized reproduction will by-pass the humble, old mechanical-and-chemical film. But for now and the next several years, film remains a strong and important medium. We'll compare what's possible there and in video in the next column. ■

This amp isn't getting a lot of attention...it doesn't need it.

The Crown PSA-2 is the most reliable high-power professional power amplifier you can buy at any price. One reviewer described it as "indestructible," since the PSA-2 accepts seemingly impossible loads, and unlike many other amp designs, will continue to produce useable output power.

PROOF OF PERFORMANCE

Crown engineers devised a simple test to show how the PSA-2 operates. Two flat metal bars are wired into the output circuit of one channel of the amp, with a music signal input. The other input is connected to a 1½ volt battery, requiring the PSA-2 to deal also with a DC signal. A heavy round steel bar is laid across the speaker leads. The amp *continues* to produce useable power, and the metal bar becomes a transducer, producing small sounds from the output signal!

ON-BOARD COMPUTING

The PSA-2 uses its built-in computer logic and unique sensing systems to determine the limits of the safe operating area of the output transistors. The PSA-2 does not just thermal out or shut down as other designs tend to do under strange loads.

It computes the level of output power at which it can continue to operate, and then orders itself to do that.

Under normal load conditions, this on-board computing makes it possible for the PSA-2 to use its output transistors more efficiently than any other protection system. There are no arbitrary cut-off points, but a continuous computing of the conditions of the output devices, and an adjustment of output to the maximum comfortable level for the amp.

NEW CONVENIENCE

Versions of the PSA-2 are now available with a choice of front and rear panel configurations. Users can select a model with on/off LED indicators for overload, signal and standby; or they can select the version equipped with the Crown "Dynamic Range Indicator," an LED array that displays peak/hold and instantaneous output for both channels. For the rear panel, a balanced input module (including variable gain and switchable hi/low Butterworth filters) is available, or unbalanced input only.

FIELD TESTED

The Crown PSA-2 amp has already proven its ability to require very little attention after it's been set up in the field. The PSA-2 has provided trouble-free power for several major touring setups, and it is now installed in a number of top-flight recording studios and auditoriums.*

TOP QUALITY SOUND

A bonus for users of the PSA-2 is its excellent sound quality. The amp proves that sonic quality, ruggedness and reliability *can* be combined in one amp.

If you haven't already considered the PSA-2 for the systems you are currently designing, write to Crown today. We'll send you a fully descriptive brochure and reprints of several reviews. They're worth your full attention.

*Names on request.

crown

1718 W. Mishawaka Road, Elkhart, Indiana 46514

Innovation. High technology. American. That's Crown.

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An Introduction to Digital Audio

WELCOME

I would like to bid you all a hearty welcome to this new series on digital audio technology. Before actually beginning, I would like to outline some of my philosophy and goals in this series. First of all, the articles during the first year will be aimed at those who have very little experience with digital signals. Secondly, I consider it more important to have a real feel for the subject rather than to have a complete mathematical treatment. When one has a feel for a subject matter, one can use intuition very successfully. Thirdly, I would like all of my readers to understand both the basics and some of the important subtleties. To achieve this goal, I am going to make many of my explanations using analogies. Your comments about the success of this approach are most welcome.

FOREIGN LANGUAGES

For an engineer who has lived all his professional life with analog audio, the learning of digitized audio is very much like learning a foreign language. The message may be the same but the grammar and vocabulary are very different. One of the interesting byproducts of learning a foreign language is a much deeper understanding of one's own native language. (A fish can only begin to truly understand water when he encounters air for the first time.) We will find this to be the case with analog signals, as we learn more about digital. For this reason, the real nature of continuum (analog) signals needs to be reviewed.

Although the languages of analog and digital signals both describe audio music signals, each has certain advantages and disadvantages. In the same way, certain languages can be better used for certain kinds of descriptions. Some poems are very difficult to translate. Also, certain

ambiguities can exist in one language but not in another. Puns, for example, usually cannot be translated. If you keep this in mind, it will be easier to understand the real differences between our comfortable continuum world of analog signals and the strange new world of discrete digital signals.

QUANTIZED VS. CONTINUOUS

Let us first begin by reviewing the vocabulary and grammar of our continuum language. An analog signal is one which can take on any value along a specified range. Thus, a current of 1.2367543 mA. is really a different current than 1.2367544 mA. Although we may not care about such a trivial difference, still, there *is* a difference. Theoretically, there is no limit to the amount of resolution which can be applied to a continuum measure. Distance can be measured in light years or in atomic particles.

In the following discussion, let us consider our "signal" to be the amount of water in a one-quart jar. We might thus have a jar with 0.78467 quarts of water. If we tried to store our signal, we might later find that evaporation had decreased the amount of water and thus degraded our signal. If we tried to pour our signal from one jar to another, the result would be slightly in error because some water would be left behind. To continue our analogy, we would say that the storage or transmission of the signal resulted in some "noise." In fact, the signal is very fragile and it could easily be destroyed. Once we have lost some water, we can never recover the amount.

Now let us consider a jar which has the capacity to contain 1000 marbles. Our signal might thus be 147 marbles in a given jar. We can store the marbles and we can pour them from jar to jar. Other than loosing a "full" marble, there will never be any error. Our signal remains perfect and there is no natural mechanism for degradation. However, unlike the jar of water, our jar of marbles is limited to one out of 1000 possible values. Clearly, the jar of water could contain more signal information since there is an infinite number of different quantities which could be in the jar. One jar could contain 0.786542 quarts and the other 0.786543 quarts. These are two different signal values. The major differences between the analog and digital languages is the following:

Rule 1. Analog signals contain more information, but that information is easily lost. Digital signals contain less information, but that information is very robust.

Since our goal is to represent audio signals in either of the two worlds, we need a mechanism of conversion between them. Using our analogy, we define one quart to be equal to 1000 marbles. This gives an exchange rate of 0.001 quarts per marble. A jar with 0.147 quarts is the "same" signal as a jar with 147 marbles.

But we are now faced with the difficult question, how many marbles are equivalent to 0.1471876 quarts? The best *approximation* is 147 marbles since we do not allow fractions of marbles. This is because the grammar of the digital language does not recognize anything but countable elements with no fractions.

Rule 2. A discrete world uses only whole numbers for counting, and all fractions must be discarded by rounding or truncation (approximations).

The fractional part which is discarded is called an error of quantization: Quantization is the act of converting a continuum signal into a discrete signal. This error cannot be recovered when we convert back again. In fact, the best approximation for all jars of water containing amounts from 0.146500 to 0.14799 quarts is 147 marbles. This form of approximation is called rounding, in that the error is no greater than ±0.0005 quarts or ±0.5 marbles. Alternatively, we could have used truncation, which is just throwing the fractional part away. This algorithm makes 0.147000 and 0.147999 quarts equivalent to 147 marbles. In one case the error is both positive and negative (rounding) and in

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db New Products & Services

STUDIO MONITOR

• The Sentry 100 is a highly efficient speaker, which while lowering distortion, allows itself to produce the sound pressure levels of comparable units with far less amplifier power. The monitor also delivers the sonic accuracy required in critical monitoring applications. An optional hardware kit for rack and wall mounting is also available.
Mfr: Electro-Voice, Inc.
Price: $200 ea.
Circle 50 on Reader Service Card

SYSTEM PREAMP

• The System Preamp has been designed to provide comprehensive and flexible control in any home stereo system, with exceptional sonic integrity. The unit lets the user route two simultaneous signal sources independently to a monitor, tapes output, or power amp and speakers. A mix control blends the two signals and permits fading from one source to another.
Mfr: MXR Innovations, Inc.
Circle 51 on Reader Service Card

MONITORS

• The Models SM-12 and SM-15 floor monitors are two-way systems designed to offer an enlightened combination of accuracy, compact size, and efficiency. A notable feature of the floor monitors is the outstanding accuracy as exemplified by flat frequency response with no peaking in the midrange.
Mfr: Cerwin-Vega
Price: SM-12 $324.
SM-15 $415.
Circle 52 on Reader Service Card

BAG END CABINETS

• Featuring a modular concept with over 40 different models, the Bag End line of cabinets allows wide versatility in matching the system to most any application. Uses range from sound reinforcement to studio to professional musical instruments. Cabinets are constructed from top grade 13-ply birch plywood with dark walnut oil finish. All hardware is flush mounted to allow for easy stacking, transporting, and storage. A 9-ply birch cover is provided for storage and handling. Machined aluminum speaker mounting hardware and expanded steel grills securely hold and protect the speaker driver, yet are designed for quick, easy access for driver replacement. Enclosures come loaded and pretested with JBL or Gauss drivers.
Mfr: Modular Sound Systems, Inc.
Circle 53 on Reader Service Card

MICROAMP SERIES

• The MicroAmp Series of precision audio amplifiers includes the L-1000 Dual Line Amplifier, the M-1000 Dual Microphone Amplifier and the P-1000 Stereo Turntable Amplifier. Available with transformers or balanced differential outputs, all models over +22 dBm output levels with extremely low noise and distortion. The latest addition to the line, the MA-1000 Stereo 10 W / Switchable Mono Bridge 25 W Amplifier is an excellent headphone or small speaker driver. Designed around the latest high slew rate integrated circuits, the MicroAmp Series provides recording studio performance in a rugged, RF protected, miniature broadcast package.
Mfr: Audio Technologies, Inc.
Circle 54 on Reader Service Card

AUDIO ANALYZER

• The Model 8903A Audio Analyzer combines a low distortion audio source with a highly flexible analyzer in one instrument which measures dc volts, ac volts, distortion, signal-to-noise, SINAD (signal-to-noise and distortion) and audio frequency from 20 Hz to 100 Hz. The new analyzer makes complicated audio measurements with a single-keystroke and is designed for use in audio test and transceiver test applications.
Mfr: Hewlett-Packard
Price: $5800.
Circle 55 on Reader Service Card

ROCK SPEAKERS

• A new line of speakers designed to reproduce the excitement and energy of rock music were recently introduced. The Rock Monitors, in order to withstand the rigors of this strenuous duty come equipped with triple protection. All three models feature a tweeter overload protection system, a circuit breaker for the entire speaker and a five-year transferable warranty. The Rock Monitors come in three sizes. They feature an 8-inch, 10-inch, or 12-inch woofer in a floor standing tower.
Mfr: Black Max Systems Inc.
Circle 56 on Reader Service Card

PORTABLE AUDIO CONSOLE

• The Sport III Dial-Up sports audio console allows the sportscaster to plug into any telephone jack and dial his station directly. An electronic rotary dial uses a twelve button touch pad to create dial pulses for the standard telephone systems virtually anywhere. A hybrid output stage gives the Sport III full talk-back capability. The unit has a built-in 2600 Hz. filter to avoid accidental disconnects and a line drop indicator to advise the sportcaster when he's lost his connection. The Sport III is also available without the dial option at lower cost. However, the dial option is plug-in and can be added any time.
Mfr: Micro-Trak Corporation
Price: $569.
Circle 57 on Reader Service Card

MICROPHONE

• The SM63-CN is a new dynamic, omni-directional type microphone with an output that is about 6db higher than larger, comparable units. The new unit is less than six inches long and weighs only 2.8 oz. Other performance features include a controlled low-frequency roll-off to insure natural sounding voice and music pickup, as well as smooth high frequency response for an overall clear clean sound similar to some condenser microphones. Included with the SM63-CN is a swivel adapter windscreen and professional three-pin audio connectors on both ends of the cable.
Mfr: Shure Brothers Inc.
Price: $100.
Circle 58 on Reader Service Card

ANALYZER/OSCILLATOR

• The Model AA 501 Automatic Distortion Analyzer and SG 505 Oscillator System dramatically lowers the cost of measuring signal distortion by reducing measurement time. Total harmonic distortion measurements can also be made quickly and without operator assistance because the AA 501's internal circuitry automatically performs level setting, tuning, and nulling functions. The high resolution, 3½ digit led display reads out distortion in per cent or dB (autoranging). An option allows measurement of intermodulation distortion on signals conforming to *SMPTE, DIN*, or *CCIF* signals. Packaged as plug-ins for the TM 500 family of modular test and measurements instruments, the AA 501/SG 505 can be combined with the user's choice of over 40 plug-in instruments in a single package. Modularity also permits remote testing.
Mfr: Tektronix
Circle 59 on Reader Service Card

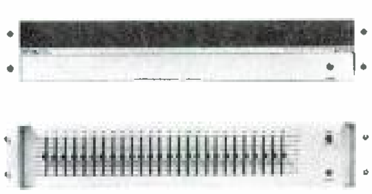
MAINTENANCE KIT

• Designed with electro-mechanical applications in mind, the JTK-79 Maintenance Kit contains a select assortment of tools for a wide variety of tasks in a 6x9x1 ¾ inch padded zipper case for easy portability and storage.
Mfr: Jensen Tools Inc.
Circle 60 on Reader Service Card

CHAMBER SYNTHESIZER

• The XL-305T "Acoustic Chamber Synthesizer" is designed specifically for broadcast applications. The reverberation system is said to provide enhancement of on-line audio signals, as well as, superior sound qualities for productions programming applications. The XL-305T includes such features as front panel selection of either monaural or stereo operation, extensive RFI protection, 600 ohm transformer signal connections and an integral, wide band mixing amplifier for on-line applications. The unit does not employ any internal limiters to accomplish this performance capability, which is based on a new design approach to reverberation. A four-band peak dip equalization section is provided for each channel that effects only the sound of the reverberation, thus allowing the user to tailor the sound of the reverb without affecting the direct signal.
Mfr: Micmix Audio Products, Inc.
Circle 61 on Reader Service Card

GRAPHIC EQUALIZER

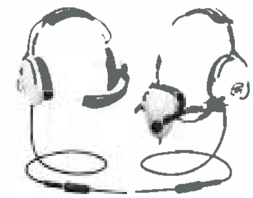


• The LR-1006 1/3 octave graphic equalizer employs 26 infinite gain multiple feedback active bandpass filters with optimally selected "Q" for proper combining with adjacent filters. Low distortion and low noise are provided by use of high slew rate low noise quad operational amplifiers. Close tolerance high quality components are used to maintain proper filter centering on ISO (International Standards Organization) frequencies.
Mfr: McMartin Industries, Inc.
Circle 62 on Reader Service Card

SOUND-LEVEL METER CALIBRATOR

• The Model 896 Calibrator is a new portable with five standard frequencies for accurately calibrating sound-level meters, and other similar instruments with A, B or C weighted relative responses. When acoustically coupled to a sound-level meter, the Model 896 provides constant sound pressure level of 114 dB or 94 dB at 125, 250, 500, 1000, and 2000 Hz at the microphone of the meter. Accuracy is better than ± 1 dB and frequency drift is held to ± 2 percent at reference conditions.
Mfr: Simpson Electric Company
Price: $165.
Circle 63 on Reader Service Card

COMMUNICATION SYSTEMS



• Sound (voice) Powered Communications Systems that provide clear voice transmissions for miles, yet require no electricity, batteries or other power sources, are available on the market of recent. Ideal for all high noise areas without power sources and applications where portable communications systems are necessary, the sound powered system features noise attenuating headsets that can be worn with all kinds of safety head gear. The headsets come with an acoustic mike boom for hands-free operation or with a noise-reducing, shielded microphone with a push-to-talk switch for use in high noise level areas.
Mfr: David Clark Company
Circle 64 on Reader Service Card

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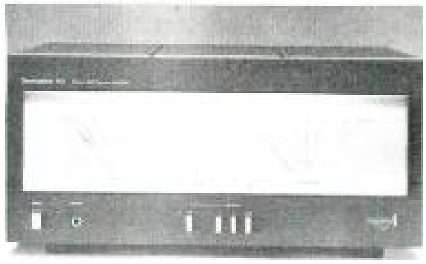
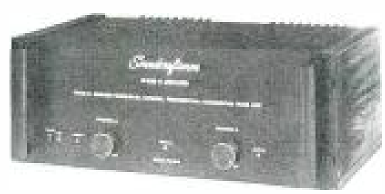
AMPLIFIER



• The new SE-A3 Stereo DC Power Amp is the successor to the SE-A1 class A+ power amplifier introduced two years ago. The SE-A3 which features the new class A circuitry, offers both the distortion-free fidelity of class A amplifiers and the high power efficiency of class B amplifiers. Major specifications are: rated power output of 200 w/rms per channel driven into 8 ohms with no more than 0.002 per cent thd from 20Hz to 20kHz; while maintaining a signal to noise ratio of 123dB. Contributing to the excellent specifications are the new class A circuitry with synchro bias that eliminates the time delay that appears as crossover distortion, independent left and right channel power transformers, low-inductance power supply capacitors, the concentrated power block to minimize electromagnetic induction, and a dual linear power transistor. The SE-A3 can drive two sets of speaker systems and is equipped with fast-response peak meters (0.01w to 300w range).
Mfr: Panasonic
Circle 66 on Reader Service Card

AMPLIFIER



• The new 1-horsepower mono dual channel basic amp was recently made available. The 750 watts at 8-ohms output of the RA7501 is rated 20-20kHz at less than .09 percent thd in the bridged mono mode and 375 watts/channel driven into 4-ohms (250 watts channel into 8-ohms) in the dual channel mode. While all performance data is at state-of-the-art level, the new amp's physical construction was specifically designed to enable its use by professional touring groups under punishing conditions.
Mfr: Soundcraftsmen
Price: $799.
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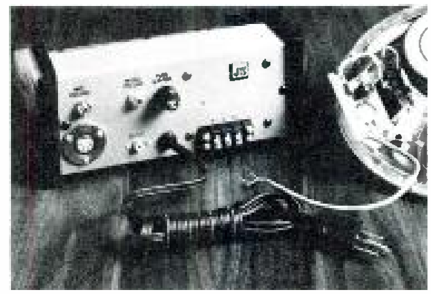
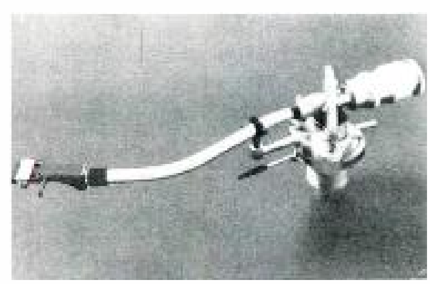
DISTRIBUTED SOUND SYSTEM



• Production has begun on the SBA Distributed Sound System specifically designed to meet the unique requirements of background music and paging applications. The system, Signal Based Amplification (SBA), makes it possible to infinitely enlarge by interrelated networking, instead of a mere addition of speakers fed by a large amplifier or a version of same. These systems consist of three components: the SBA-PM (master), the SBA-R (loudspeaker remote with built-in volume control), and, if needed in larger installations, the SBA-P (supplementary) power unit. The SBA system utilizes a special signal containing the audio program plus a precisely generated bias voltage determined directly from the program signal and varying continuously with the level of the audio program. Frequency response is 10-20,000 Hz and is achieved by elimination of line transformers through direct coupled amplification.
Mfr: J. W. Davis & Company
Circle 68 on Reader Service Card

TONEARM



• The Model EPA-100 is a variable dynamic damping universal tonearm with ruby, ball bearings. The EPA-100 represents a number of significant advances in tonearm technology: user's ability to "tune" the arm for cartridge compatibility via the variable dynamic damping system, significant decrease in static friction in a gimbal 4-point suspension, and an extremely lightweight tonearm by using titanium nitride.
Mfr: Panasonic
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db Editorial

Radio and Recording— Part II

LAST MONTH'S COVER gave us a beautiful view of the Saul Zaentz Film Center, and a good look at the work of noted acoustical consultant Jeff Cooper. As we chatted with Cooper at the recent AES convention, it soon became apparent that, in addition to the cover, there was a good story idea here as well.

Need we say more? Cooper's work at the film center is our lead feature this month.

If you're not quite ready to build your own film center, perhaps you'll settle for a PRASP. As author Scott Pelking notes, "...it's cheap, and it works." Of course, the moveable-slat idea is not a new one: studio architects have been designing them into "super studios" for quite some time. But such studio projects are far beyond the budget restrictions of many of us. So, if you're considering a little do-it-yourself studio construction in the near future, why not consider a few PRASPs? What's a PRASP? Turn the pages and find out.

If you *are* able to call in the experts, you'll find a brief listing of some of them in our Directory of Studio Construction Services. When we first thought about publishing such a directory ages ago (at least, it *seems* like it was ages ago), we thought it was one of our more-clever ideas. Now, we're not so sure anymore. For instance, who (or what) is a studio construction service anyway?

Some of the best-known consultants in architectural acoustics do not provide their own in-house construction services. As George Augspurger noted last month, he prefers to work in cooperation with an architect or studio supply house. (See Recording Studio Design and Acoustics—Ed.)

On the other hand, the proprietor of the local hi-fi hardware store has an uncle working in the stock room who always wanted to build a studio for someone. You know the rest of this sad story....

And then, there are the first-rate turnkey operations with architects and crew on staff. Others can arrange such services on an "as-requested" basis. And so it goes.

Throwing caution to the winds, we list a handful of well-known companies. Some offer blueprints and supervision only. Others have their own tool boxes ready and waiting. Still others will not only build your studio, they'll help you stock it too. Before proceeding, make sure you have a clear understanding of what you need, and what you may expect to receive from the people who are helping you.

And if you need help with your hardware list, have a look at our AES Convention Report in this issue. Once again, digital audio is in the news, but as the technology gets more sophisticated, there seems to be a counter-current flowing in the direction of simplicity. This will certainly not wash digital out-to-sea, but it is refreshing to watch (and hear) people re-inventing the (audio) wheel. The latest discovery seems to be the stereo microphone, which was previously discovered more than 50 years ago. Try it; you may like it!

In our London AES Convention Report (May, 1980), we promised more details about the "Golden Ear" tests conducted by KEF Electronics, Ltd. European correspondent John Borwick looked in at KEF for an update on the "Great (Golden Ear) Debate." And while he was at it, he had a closer look at some of the more advanced testing procedures, which may no longer be ignored by professional studio engineers.

DIGITAL AUDIO AND SOUND WITH IMAGES

Beginning this month, we welcome Dr. Barry Blesser to our list of columnists. Dr. Blesser is a fellow of the AES, and the author of "Digitization of Audio: A Comprehensive Examination of Theory, Implementation, and Current Practice," which appeared in the October, 1978 Journal of the Audio Engineering Society. In his spare time, he invented the PDM dynamic range compression system, and developed the first commercially-successful digital delay system—the Lexicon DELTA-T model 101, which was marketed in 1970 by Gotham Audio.

In short, Dr. Blesser knows just a little something about digital audio and now he shares his knowledge with us. Come on along, as we learn more about Digital Audio, by Barry Blesser.

Also starting this month, Neal Weinstock shares Sound With Images with regular columnist Martin Dickstein. As sound becomes more important to image-makers, we expand our coverage to include the world of video tape, the video disc, and motion picture film. In fact, even the television industry is beginning to discover audio. Stay with us, as Dickstein and Weinstock keep us up-to-date on the important side of pro audio. ■

JEFF COOPER

The Construction of the Saul Zaentz Film Center

Formerly known as Fantasy Films, the Saul Zaentz Film Center is a prize example of an emerging phenomenon in the entertainment business—the alliance of film mixing and music recording.

BACKGROUND

This story begins in the fall of 1978 when the Saul Zaentz Company first approached our firm to act as acoustical design consultants for the construction of their new film mixing facility, to be located in Berkeley, California.

The Saul Zaentz Co. had by this time distinguished themselves as one of the San Francisco Bay area's finest film production organizations. Their past credits included such noteworthy films as "One Flew Over the Cuckoo's Nest," winner of five Academy Awards, and the animated version of Tolkien's classic, "Lord of the Rings." The Saul Zaentz Co. also gained recognition in the field of music through their highly innovative and respected **Fantasy/Prestige/Galaxy** record label.

The company's ultimate goal was an ambitious one. It centered on the construction of an 85,000 square-foot ground-up complex which would contain all the creative facilities necessary for start-to-finish record *and* film production. The new building would include facilities for film editing, music and voice recording, disc mastering, sound mixing for video, film mixing and office suites, as well as other support facilities.

PROJECT GOALS

The company was quite specific about its goals for the film mixing facility. Their requirements were as follows:

1. The facility must be capable of providing fully-computerized mixing capability for Dolby-surround feature films using up to nine discrete speaker locations. The facility must also have the flexibility for changing audio formats to accommodate future technological innovations.
2. The facility must be designed to the most exacting acoustical standards, enabling mixers to make accurate judgements of the final qualities of the mix. Most important, the final mixes must be capable of translating accurately to a wide variety of real world theaters, most of which would of course be considerably larger in volume than the mixing room itself.
3. Support facilities must be provided adjacent to the theater including a 35mm and 16mm projection room, a machine room complete with computer-access floor (for wiring of synchronized film dubbers), a film-transfer room, an electronic looping stage for dialogue replacement, a small voice-over control room and studio, and a *"roomizer"* or acoustic simulation chamber, in which various acoustical environments could be simulated for purposes of realism in film dubbing.

THE DESIGN PROBLEMS

In the original design of the seven-story building shell, the company's architects had allocated a 25 ft. x 50 ft. rectangular two-story portion of the building for the film mix facility. The main entrance was to be from the upper level at the rear of the theater, via a step-down arrangement of various cast-in-place concrete seating levels.

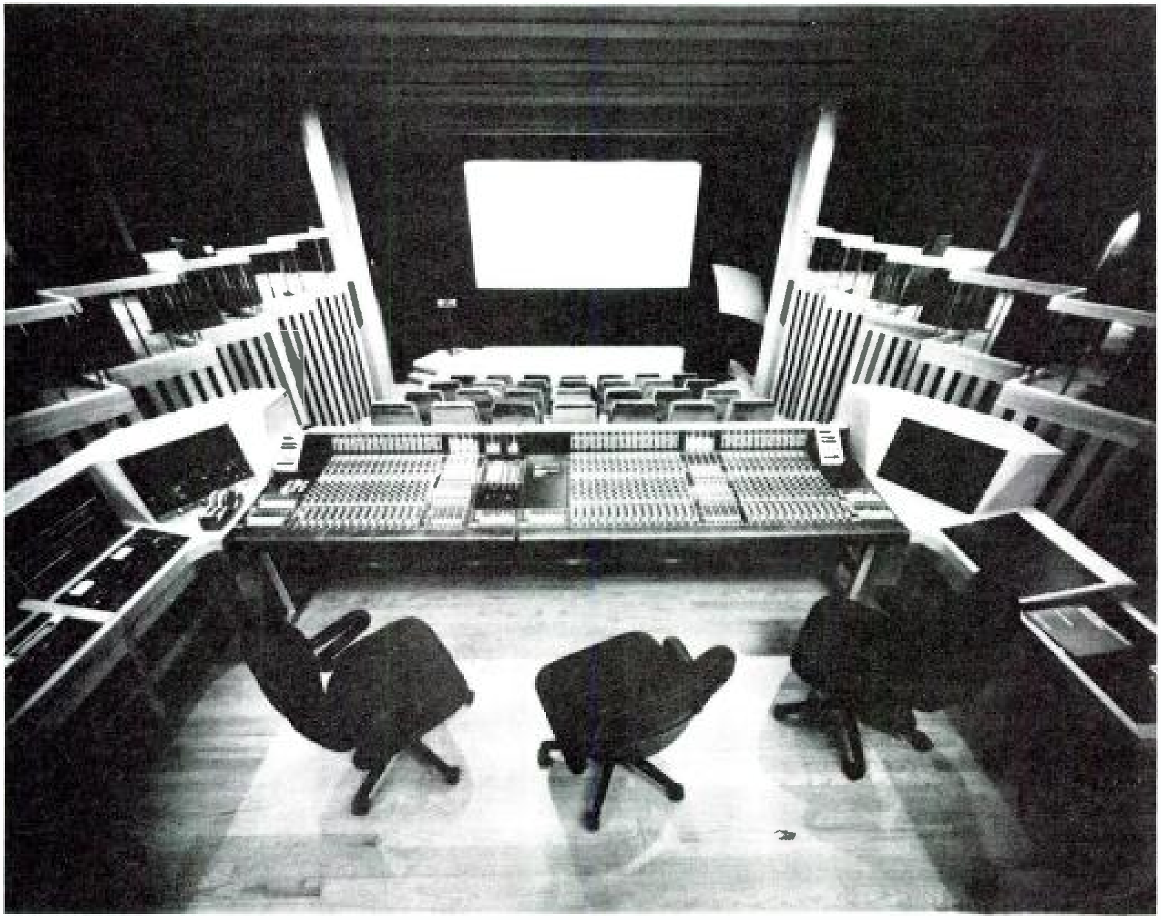
Overall width of the room was limited by the placement of internal building columns, which had already been cast on 25-foot centers.

Our first reaction was that the space allocated for the theatre was unfortunately too small for a project of such an ambitious nature. Specifically, we felt that the 25-foot width was the most restrictive parameter.

Sound tracks in Dolby-surround mixes may contain audio information in the 40 Hz range, and in some special instances go as low as 10 Hz for subsonic special effects. As a result, *axial room modes* and objectionable low frequency colorations can easily be excited, thereby destroying the acoustical objectivity of the mixing environment. Axial modes are fundamental resonances inherent in the geometry and dimensions of all rooms, ie.:

F_R = Vn/2D
F_R = resonant frequency
V = velocity of sound
n = 'enth harmonic (1, 2, 3,...n)
D = a room distortion

Jeff Cooper is an architect whose Los Angeles-based acoustical consulting firm specializes in the design of recording and film studios.



A view of the Saul Zaentz Film Center looking over a three-man computerized Harrison console and several rows of tiered seating. Design by Jeff Cooper.

Such fundamental resonances become less apparent in very large theaters, where all dimensions exceed 30-feet. This is due to the positioning of the first and second harmonics of the axial room modes below the low end roll-offs of most commercially available loudspeakers.

We felt that the axial room modes in a 25 ft. x 50 ft. rectangular theater (as the original plan intended), would have lead to coincident multiple resonances, because the length was an integral multiple of the width. These resonances would have been extremely difficult to eliminate, given even the most selective absorption treatment (ie. absorbers designed to work at specific frequency bands).

We calculated basic axial resonances for the shell, after soundproofing, at 25.5 Hz, 51 Hz, 76.5 Hz, 102 Hz, and 127.5 Hz, etc.

From our calculations, we determined that the finished volume of the theater would be about 17,900 cubic feet, and the area of the boundary surfaces (floors, walls, ceiling) would be approximately 4,600 square feet. Using the modified Sabine equation given below, reverberation time at 500 Hz was estimated at about 0.2 seconds.

$$T = \frac{.049\ V}{S \log_e(1 - \alpha) + 4\ mV}$$

T = reverberation time
V = volume
S = surface area
α = mean absorption coefficient (0.5-to-0.6 range)
m = air attenuation coefficient (less than 0.1)

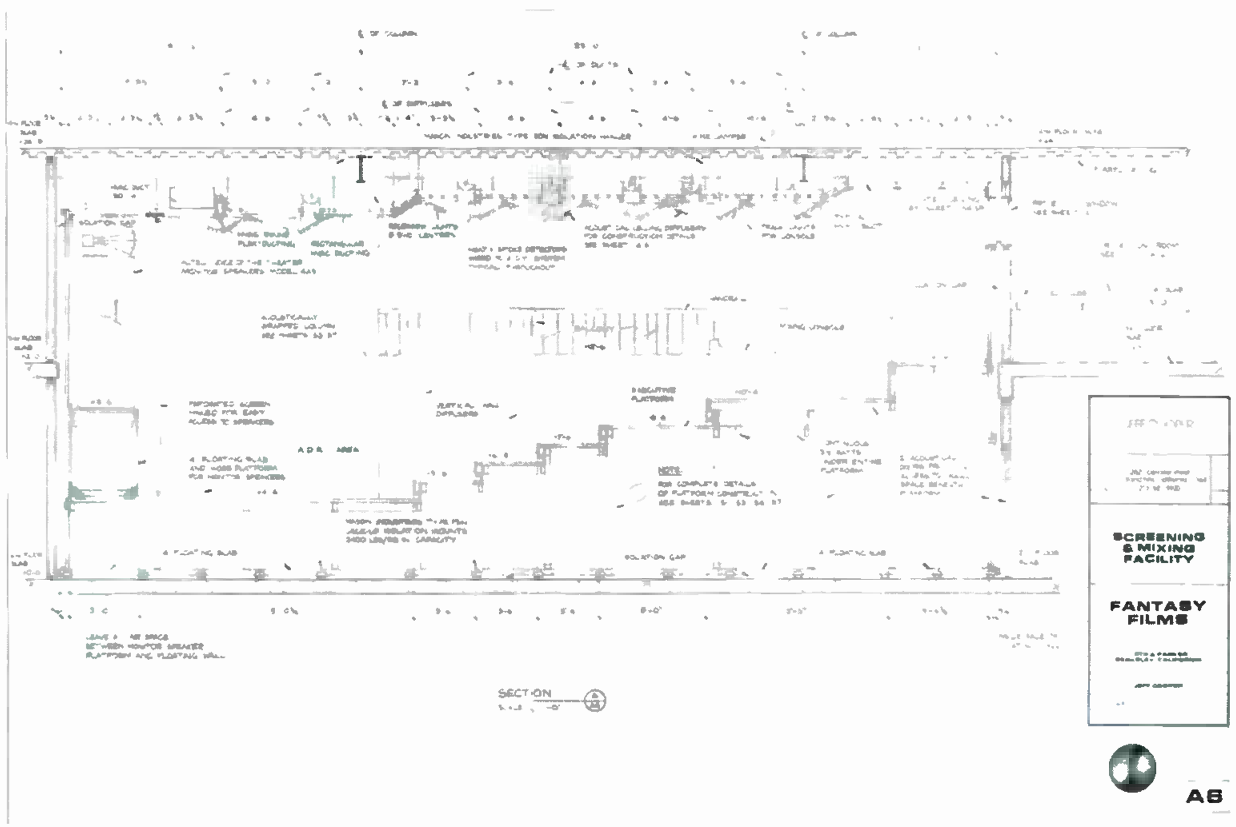
We felt that this reverberation time would be too low to approximate the larger movie theaters, which was one of the expressed goals of the facility. Screening theaters usually have reverberation times which vary from 0.3 seconds to 2.0 seconds, depending on internal volume and occupancy levels. One seated occupant provides approximately 4.75 sabins of absorptivity, a quite significant figure when multiplied by a potential audience of one thousand or more persons in larger theaters. (W. C. Sabine, *Collected Papers on Acoustics*, Harvard University Press)

RECOMMENDATIONS

Due to the above factors, we concluded that some fundamental changes in the basic layout were necessary. First, we recommended that the shape of the theater be changed from a 2:1 rectangle to a fan or horn-shaped enclosure, flaring outward from the screen. Second, we recommend that the overall width be increased to 43-feet giving a maximum width-to-length ratio of 1:1.28. Lastly, we recommend that the volume of the theater be increased by a minimum of 4,100 cubic feet to a total volume of approximately 22,200 cubic feet.

Our primary goal was to push the reverberation time into a higher range without resorting to overly-reflective room surfaces. On the other hand, we still desired a reverberation time slightly less than the previously-considered optimum of 0.8 seconds for medium sized theaters and film mixing facilities[1] in order to improve source localization and orientation for multi-channel mixes.

Our secondary goal was to create a diffuse listening environment, with an extremely-wide homogeneous sound field in the mixing area. Unlike music mixing, film mixing often requires



Cross-sectional drawing of the main theater and projection room.

three or more engineers working side-by-side in order to simultaneously mix music, dialogue, and effects. This results in the need for an accurate mixing zone in excess of 20-feet, unheard of in most recording studio control rooms.

THE REACTION

Our comments were not greeted with great enthusiasm by project architects and builders. The basic design of the building and allocations of interior floor space had already been completed, and there was no doubt that our suggestions required substantial modification of the original building plan. In fact, at the time our suggestions arrived, plans of mechanical and structural systems had already been drawn, and the erection of concrete columns and post-tensioned slabs was well under way. It was unfortunate that the architects had failed to contact us during the *early* design stages of the project—a mistake that is all-too-common in many large scale construction projects. This type of error often has disastrous results!

To complicate matters, the structural engineer's choice of *post-tensioned concrete construction* for the entire building was also disappointing. This system is extremely inflexible and allows little room for modification.

In this type of construction, reinforcing rods are placed inside the slabs at critical locations with pre-calculated stresses. After curing, the reinforcing members are tensioned to accommodate heavy structural loads through the slab. Once the slabs have been poured and post-tensioned, they can neither be cut, removed, nor penetrated in any way without upsetting the entire distribution of loads throughout the superstructure. In fact, in order to drill an ordinary hole for telephone cable, it is necessary to X-ray the slab first, to avoid hitting any reinforcing rods.

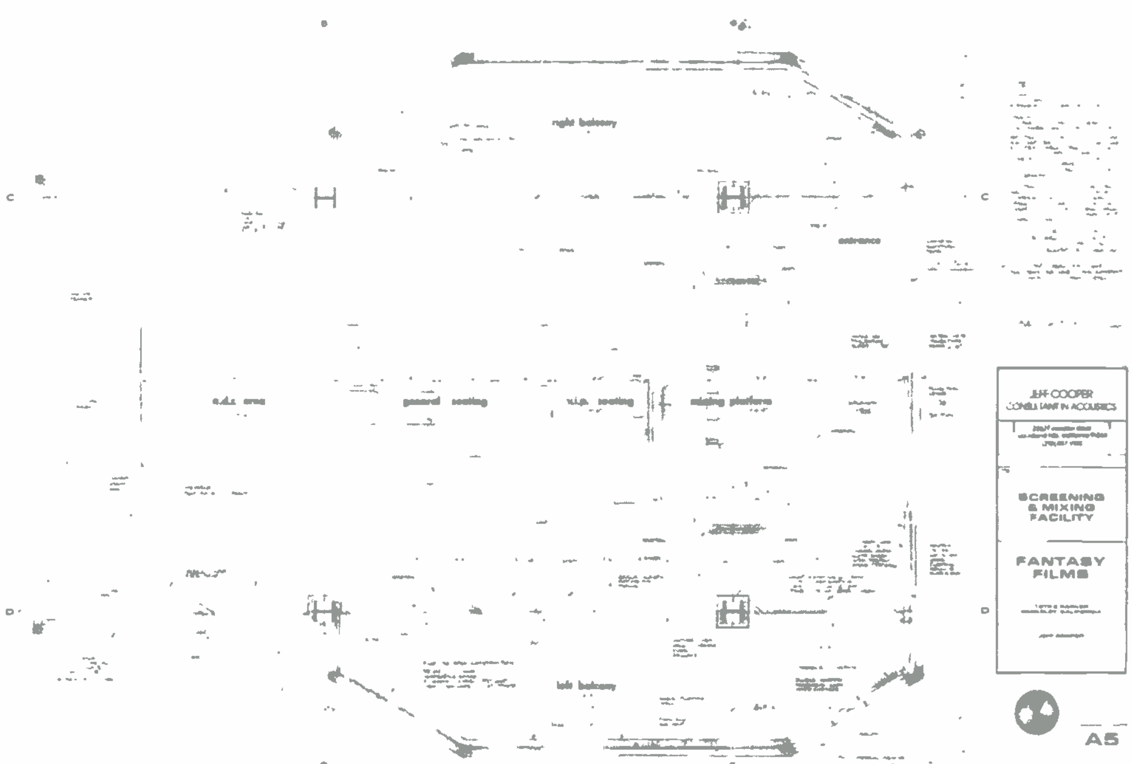
Our task was to find a way to modify the basic layout of the second and third floors of the building to achieve the goals of non-rectangular room shape and increased volume and width without fundamental change to the existing building shell.

THE SOLUTION

Since the cut-out for the original 25 ft. x 50 ft. room had already been cast, it was virtually impossible to re-shape the entire shell. Instead, we opted to leave the lower floor in its basic rectangular configuration, and to expand the upper level to the desired 43-feet, taking the space from an adjacent lounge, a hallway, and offices. The upper level was reshaped by the addition of two wings on either side, tapered toward the front of the theater. This gave us the desired horn shape discussed above. The modification increased acoustical volume by 27 percent, bringing the reverberation time up to the desired range of 0.8 seconds-1.2 seconds.

In order to achieve our goal of diffusion, large 3 ft. x 5 ft. variable height triangular diffusers were designed to scatter sound waves as low as 40 Hz. These constructions were designed to extend from the various seating levels up to the second level floating slab, where they would be architecturally capped. Since the top of each diffuser could be accessed from the second level wings of the theater, the diffusers could also be utilized as balcony extensions, capable of providing additional seating space. Each diffuser could accommodate up to two seats, a potential extra seating capacity of 16 persons. Despite the havoc that these modifications created with the original building design, the changes were greeted with great enthusiasm by the film department staff—especially the project's chief co-ordinator, Irving Saraf. Saraf felt that acoustical considerations were of paramount importance if the facility was to achieve its initial goals.

Plan of the third floor showing main theater and support areas. Note that the transfer room contains windows to both the main theater and the roomizer (acoustic simulation chamber), enabling the transfer room to be used as an ADR control area.



Plan of the main theater showing mixing platform, seating, ADR area, screen, and front monitor speakers. Acoustical-diffuser balconies are visible to the left and right of the general seating area.

A side view of the Zaentz Film Center depicting the graduated seating levels. Notice the 1 in. x 40 ft. air slots in the top of the picture.

THE FINAL DESIGN

Once the new design concept was approved, we set about the task of doing our calculations and final drawings. In order to achieve an isolation level on the order of 65 dB (500 Hz) we determined that it was necessary to provide a fully-decoupled wall and ceiling system, as well as a floating slab. Achieving this was not at all straightforward. The studio consisted of two levels and due to the modified widened design, now contained four main building columns within its shell. These columns would have to protrude directly through the floating shell.

The final solution was accomplished by floating separate 4-inch concrete slabs for each of the different levels (three slabs total), and in effect, cantilevering the upper slabs past the edge of the original slab. This enabled us to attach the upper floating slabs to the lower floating slab by using vertically isolated wallls, thereby completing the sound seal and maintaining the important intersticial air gap.

Isolating the columns posed another problem. To solve this, the lower slabs were poured to within a 1½ inch border gap of the column bases. A lead foam, and plywood shell, to be supported by the lower floating slab, was designed to encapsulate the 24-foot high columns. This effectively continued the floating shell up to the ceiling level, completing the sound seals.

THE CONSTRUCTION PROCESS

FLOATING THE SLABS

In order to decouple the floor slabs, which carried all the loads of the floating shell, the jack-up system of construction was chosen. Steel and neoprene bell-shaped isolators were placed in a grid of reinforcing rods, approximately 5-feet on center. These were separated from the original slab by only a 6-mil polyethelene vapor barrier. The new slab was poured directly on the existing slab. The isolator tops were used as a screed level.

After the new slab cured, a threaded bolt was inserted into each isolator and the slab was literally jacked-up to the desired level. Advantages of this system over other floating slab methods included a guaranteed contact-free slab with adjustable height. In this case, the required airspace was two inches. Mason Industries of Los Angeles was sub-contracted to float the slabs.

BUILDING THE INFRASTRUCTURE

After curing of the second- and third-floor slabs, a post-and-beam grid was built, to act as a structural framework for multi-tiered seating levels. We opted to build this entire infrastructure from wood, to increase flexibility and to avoid ringing and other resonant problems associated with steel construction. Due to stringent building codes, fire retardant lumber was required.

Since the mixing platform and upper seating levels were almost 12 feet higher than the floating slab, the sealed area created below the infrastructure provided a perfect access space for conduit, auxiliary equipment racks, dimmer motors, and other electrical and mechanical necessities. Penetrations for these items could be made whenever and wherever required, because they were, of course, contained *inside* the floating shell. Extensive penetration of the outside shell, which would have destroyed the integrity of the sound seal, was avoided.

Next, a floating ceiling was hung from the slab above using spring and rubber isolators on 4-foot centers. Penetrations for duct work were kept to a minimum of one-per-diffuser.

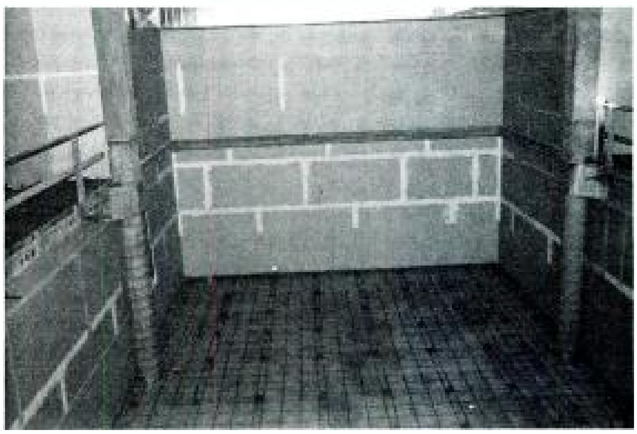
Proper air distribution was accomplished *inside* the infrastructure by utilizing 13½ inch deep sawtooth shaped ceiling membrane resonators, which had been designed to attenuate bass and to diffuse sound. These resonators were hung from the floating ceiling and the contained airspace was used as a plenum. Thin slots were cut to distribute the air evenly across the entire width of the theater. These 1 in. x 40 ft. long slots are just barely visible in the accompanying photograph of the finished facility.

It should be noted that the large cross-sectional area of the slots kept air velocities below 300 lineal ft./minute, in a range where air flow would not interfere with the low ambient noise levels required (below NC-25).

Incandescent dimmer-controlled lighting fixtures were recessed into the membrane absorbers as well. This enabled heat produced inside the light cannisters to be immediately vented by incoming cool air. Conversely, the chill factor was removed from the supply air, creating a more comfortable differential between room air and incoming air.

In the accompanying cross-sectional drawing, the floating slab, jack-up isolators, multi-leveled seating platforms, floating ceiling, membrane resonator plenum, and recessed lights, can all be seen.

View from the projection booth after application of first soundproofing wall. Jack-up isolator grid and reinforcing rods for floating slab are visible in the lower portion of the photograph. Two freshly poured floating balcony slabs are visible on the extreme left and right.



OTHER CONSIDERATIONS—AMBIENT LIGHT

In addition to low ambient noise levels, mixing theaters of this type also require notoriously-low ambient light levels. This allows accurate assessment of image quality and color correctness. In a complex with many architectural level changes, a need for low light is in conflict with safety requirements. Producers and directors must be able to come and go during a mix, and all occupants must be able to direct themselves quickly and safely to exits in case of an emergency.

To solve this dilemma, a floor-concealed indirect lighting system was developed. This system used one-inch diameter amber incandescent tubes commercially known as *lumilites*. Fluorescent tubes were not considered, because of noise-producing problems and electrical interference. The extremely-low-wattage lumilites were fitted in a regular pattern beneath steps and seating level overhangs. Brightness was controlled by a motorized dimmer, with duplicate controls in the projection room and at the mixing console.

Using this system, house lights could be turned off completely during screening without compromising safety. Low-level light was provided only at floor level through continuous 1½ inch slots (reminiscent of the air conditioning slots above). Ambient light level build-up due to the step lights was negligible.

In order to minimize ambient light build-up due to screen reflections, the absorptive walls of the theater were covered with a dense chocolate-colored light-absorbing velvet. The use of this particular fabric kept theater surfaces virtually non-reflective, while enabling the compressed fiberglass beneath to effectively perform its function as a high- and mid-frequency absorber.

CONCLUSION

Upon completion of the acoustical infrastructure, the finished materials were applied to walls, ceiling and floor. Owens Corning #703 linear glass cloth was used as a high-mid frequency absorber in wall and ceiling, and various-width solid oak slats were used as high-mid reflectors. These slats were fitted directly over the side wall membrane diffusers to permit simultaneous bass absorption and high-mid reflection in the main seating and mixing area.

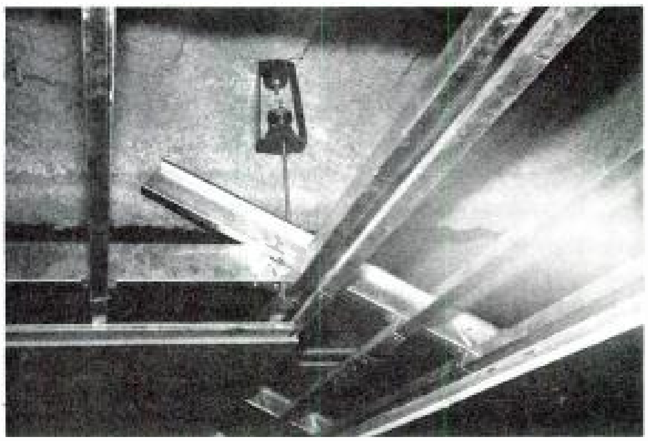
A 12 ft. x 20 ft. area at the front of the theater was treated with a hardwood floor to enable the room to also be used as a dialogue replacement studio (ie. for allowing actors to overdub dialogue while watching their own movements on screen). A small console and monitor speaker system were provided for control purposes in the transfer room adjacent to the left balcony. A 4 ft. x 7 ft. double-paned view window was provided to enable direct vision of the screen from the control booth. This window was fitted with a removable absorptive cover to restore total acoustic symmetry to the theater during mixing.

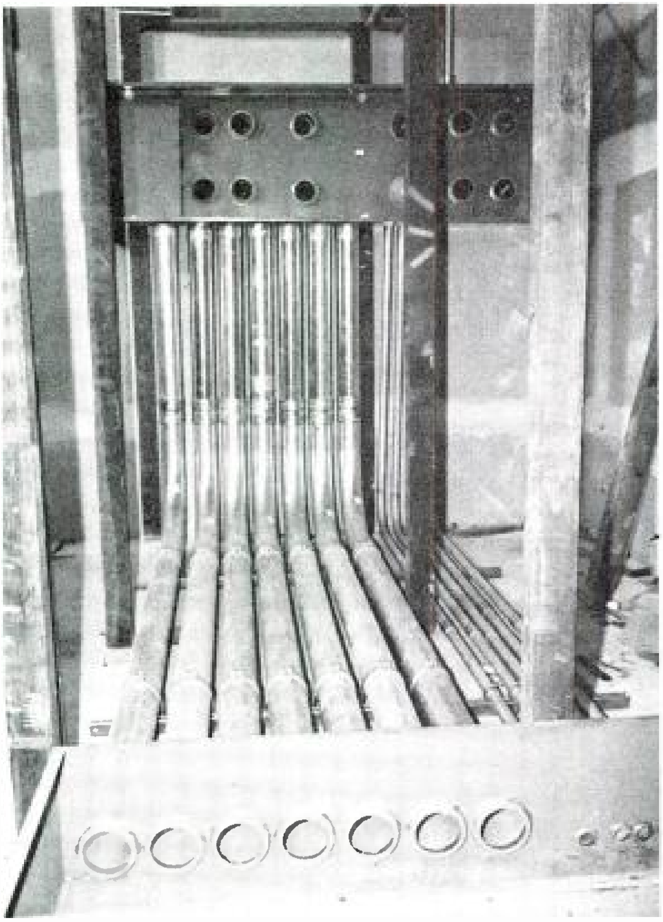
The theater was fitted with three Altec A4 cinema loudspeakers behind the perforated screen and six additional Altec A8 surround speakers on the rear and side balconies. Side, front, and rear surrounds were made movable, to allow experimentation with new audio formats.

This project represented a new type of alliance between two increasingly-related technologies: film mixing and music recording. Certainly as high fidelity multi-channel sound systems become available to screening theaters and demand for audio quality in film increases, new experimental theaters like the one described above are bound to proliferate.

New experiments in sound mixing or film, highlighted by such recent examples as "Apocalypse Now's" Academy Award winning sound assemblage, have proven that we can look forward to a continuing development of the partnership between sound and film. ■

Close-up of a spring rubber isolator typical of those used to suspend the framework for the primary floating ceiling. The three-layer sound sandwich to be applied is visible in the lower right-hand corner of the photograph.





Sampling of the many conduit runs located beneath the theater to accommodate control wiring, audio wiring, and video tie-ins to the recording studios one floor below.

REFERENCE

1. Rettinger, Michael, *Acoustic Design and Noise Control*, Chemical Publishing Co., Inc., New York (1973)

SCOTT PELKING

The PRASP—And How to Build One

The PRASP was designed to fill the absorptive/reflective void in the recording studio at Bowling Green State University. It has many practical uses. Once again, "necessity is truly the mother of invention!"

PRASP STANDS FOR PORTABLE REFLECTIVE ABSORPTIVE STUDIO PANEL. It improves acoustics during recording sessions, and may help eliminate the "egg-carton compromise" from small professional studios with tight budgets. And now for the good news: you can build it yourself inexpensively.

"NECESSITY IS THE MOTHER..."

I designed the PRASP during a recording techniques class at Bowling Green State University, taught by independent recording engineer David Lau (Ann Arbor, MI). It was the first of its kind at the University; just that fall, the recording/electronic music studio complex opened for classes, even though much of the equipment had yet to arrive and be installed.

The recording studio had only two large-area flip panels, both of which were mounted on the same wall. About midterm, we discovered after some percussion and horn sessions that the panels were not doing much to alter the room's reverberation characteristics.

What to do? We set up some rolls of carpeting along the walls, securing them to the walls with duct tape. We even propped them up with classroom chairs. Though this worked fairly well, it did create some problems: when the tape was removed, it robbed the wall of a patch of paint. If a musician or technician accidentally bumped the carpeting, the whole mess was liable to collapse. And, the set-up crews often stabbed their fingers on well-hidden staples.

Those of us who usually volunteered to prepare the studio for recording sessions began to grumble. Dave Lau sympathized, but he wasn't too sure what to do about it. He thought about maybe mounting the carpeting on panels, if one of the students wanted to do the work. While doodling in my notebook, I scribbled out the first ideas for what would become the PRASP. Later, I built the first prototype, and then, two more PRASP's. All three units are currently mounted in the College of Musical Arts recording studio (Bowling Green, Ohio).

CONSTRUCTION BEGINS

Basically, the PRASP consists of a wooden frame housing nine pivoting wooden slats. The reflective side of each slat is merely finished wood; the other side has thick pile or shag carpeting tacked and glued to it. Hanger bolts fit into the butt ends of the slats and into corresponding holes in the horizontal braces. Lubricated washers fit between the ends of the slats and the underside of one horizontal brace, and between the face of the horizontal brace and the securing nuts. Hex nuts are used on the top-end bolts; wing nuts are used on the bottom end, to allow us to secure the slat at any angle, and to loosen the slat to move it.

As in the big-budget studio, the reasoning behind the pivoting slats is to create varying degrees of reflection or

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Scott Pelking is a freelance author and a student at Bowling Green State University.

absorption. The reflected waves can be angled to the right or left, or both. Before it was mounted, one student placed a guitar amp behind the PRASP, the speaker facing a solid "absorptive" wall, with one slat angled enough to poke a microphone through; he managed to record the electric guitar and an acoustic piano at the same time with (he assures me) good results.

MATERIALS

The completed PRASP measures 68½-inches wide by 48-inches high by 7¼-inches deep. Each slat measures 46-inches long, 7¼-inches wide, and ¾-inches thick. Using the same-sized stock throughout is not a hard-and-fast rule; it should not be difficult to alter the design to fit the needs of the individual builder. I used all the same stock—96-inch long by 7¼-inch wide by ¾-inch thick pine—it was simpler than buying different-sized stock. Most of the boards looked similar, so I figured little discrepancy would develop among the pieces once they were stained and finished. It is a good idea to buy the straightest stock possible, but a slight warp can be tolerated.

Buy the least expensive carpeting. Ends-of-rolls and remnants usually sell more cheaply than plentiful, in-stock items. It may be possible to use some old carpeting lying around in the attic or a corner of the garage. At any rate, use thick carpet rated Class I, II, A, or B, depending on your state rating system. Not only will this carpet be more absorptive, but it will also comply with state fire code sections concerning "carpeting on walls."

Other materials include:

8—standard 2-inch corner brackets
12—¾-inch long wood screws
18—¼-inch x 2½-inch or ¼-inch x 3-inch hanger bolts
36—wide ¼-inch washers
9—¼-inch hex nuts
9—¼ wing nuts (or 18 wing nuts and no hex nuts)
2—7½-inch x 9-inch L-shaped mounting brackets
2—5-inch x 7½-inch L-shaped mounting brackets
12—anchor screws (don't use ordinary wood screws—they'll pull out)
packet of carpet tacks
bottle of good quality glue
household utility oil
can of stain/varnish or a can of stain and a can of clear top-coat finish
can of turpentine or paint thinner

The following tools will also be necessary:

safety glasses
saw
power drill
tape measure
carpenter's square or good metal ruler
box-end or adjustable jaw (crescent) wrench for hex nuts
hammer
router with ¾-inch rabbet-joint bit or appropriate chisels
screwdrivers
tack cloth
wide paint brush
old, clean, lint-free rags
sandpaper (medium, fine and extra-fine or 600-grit finishing paper)

If possible, use a table saw or circular saw to cut the stock to size; the pieces will be more uniform and the edges truer. A sander and a buffer, or a sander/buffer jig for an electric drill, will save a good deal of time in the sanding-and-finishing stages. A drill press, or a press jig for a portable drill, is handy for truer hole alignment. A Black and Decker "Workmate" is handy for securing the stock while sanding, finishing, sawing, and cutting carpet. Many rental establishments have power tools available. If you're not an experienced carpenter *learn about power tools before using them! Wear safety glasses while working!* (It's very difficult to read meters without eyes, and next-to-impossible to adjust knobs and switches without fingers.)

BUILDING PROCEDURE

Cut all stock to size. If you use dimensions different from those given, remember to make allowances for pivot radii and joint depth.

Cut ¾-inch wide rabbet joints on the underside ends of the horizontal braces, at a depth of a little less than ⅜-inches. If a straight-cutting jig for the router is not available, nail a true-edged piece of scrap lumber so that the flat edge faces the end of the brace. Measure the distance by placing the router on the near edge of the underside of the brace, its blade just touching the width mark. Position the piece of wood and nail it down, using thin nails or brads, then make the cut. Carefully remove the block and reposition it to cut the next joint.

Measure, mark off, and rout out the corner brack inserts. These should be as deep as the brackets are thick, and slightly larger than the bracket's surface area. These inserts should be made only on the horizontal braces, since the end slats will have enough room to pass freely over the vertical brace segments of the brackets. A template of cardboard and masking tape should make this particular job a little easier.

Measure off and mark pivot holes on the horizontal braces. Be sure to allow ½-inch between the edge of the slat and the inside face of the vertical brace (measured from the inside edge of the rabbet joint.) The pivot holes should be forward of center by ½-¾ inches, to ensure that the slats will be unhindered by the rear wall or any back-braces. If possible, clamp the two horizontal braces together and drill through both *at the same time*. This guarantees true correspondence of pivot holes, and requires making only one piece of wood. Use a ¼-inch drill bit.

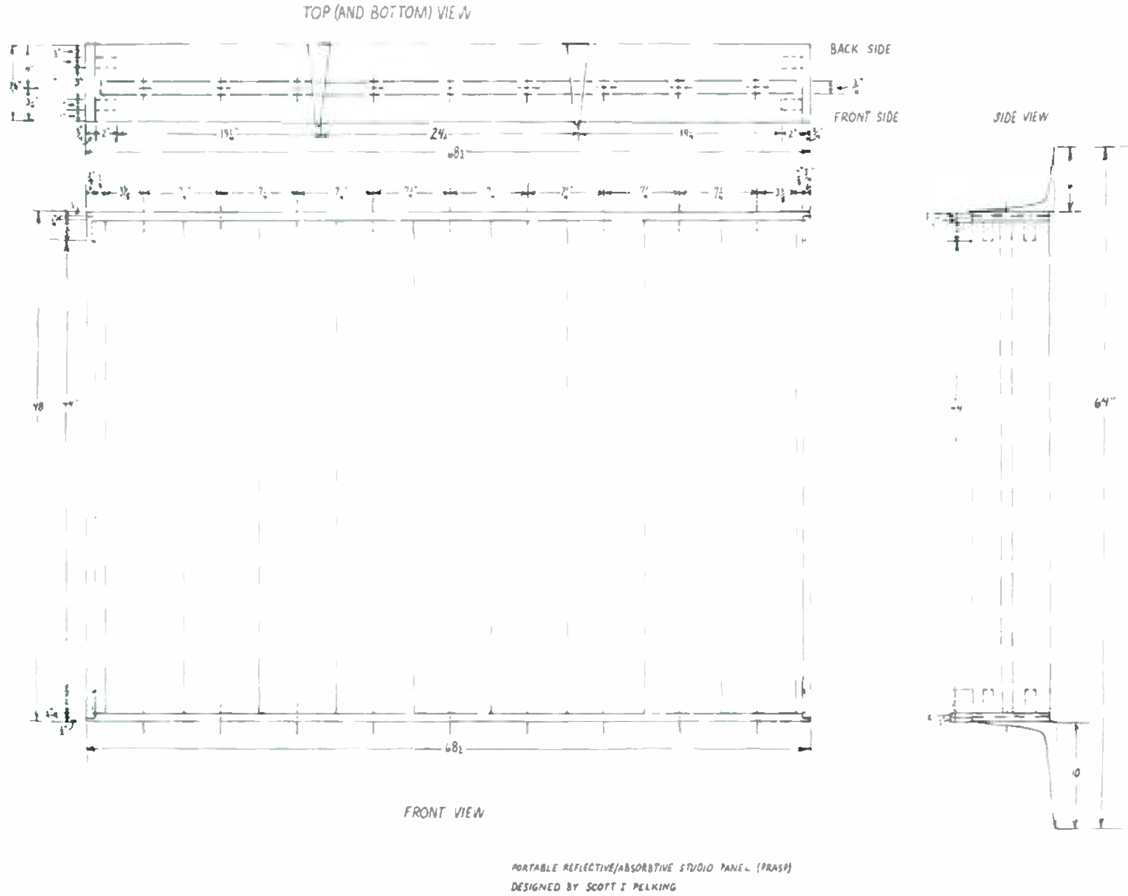
The spacing of the mounting brackets is probably the trickiest and most important aspect of mounting the PRASP. Using the 7½-inch side of a bracket, mark the exact places where the mounting screws will go. The brackets should be no more than 24 inches from the edges of the horizontal braces nor less than 20 inches. Use the larger brackets on the bottom brace, the smaller on the top. In either case, secure the 7½-inch side to the wood.

The easiest way to place the anchor-screw pilot holes in the wall is to mark the wall with a sharp pencil while a friend holds the bottom brace *level*. If you don't have a carpenter's level, use a clear glass of water. Be sure it's level; take the time to be sure so the mounted PRASP won't lean. Insert the plastic anchors into the holes. Detach the mounts from the bottom horizontal brace and attach them to the wall.

Sand faces, sides, and ends of all pieces. Use medium grade sandpaper first (unless the wood is very rough; then, start out with coarse). Once the wood is fairly smooth to the touch, wipe off saw dust and begin sanding with the next finer grade of sandpaper. Depending on how concerned you are with the quality of the finished wood, you may then remove all scratches by hand-sanding (or by using a finishing sander) with extra-fine or 600-grit finishing paper.

Cut carpet to roughly the size of the face of the slat, using a single-edged razor blade. I have found it simpler to glue and tack the section of carpet on the slat first, and trim the excess afterwards. Spread a thin layer of glue on the face of the slat, then place the carpet segment, back first onto the glue. It may be beneficial to clamp the carpet down to get a good seal, but it is not crucial. Hammer tacks into the carpet every two inches or so, about ½-inch from the edges and down the center.

Carefully measure and mark the *exact center* in the ends of each slat. Using a ⅛-inch or 3/16-inch drill bit, bore a pilot hole *as straight as possible*. Once the pilot holes are made, screw in the hanger bolts by inserting the *bolt end* (the blunt end) into the chuck of the drill and tightening the chuck as you would with a drill bit; place the *screw end* (the sharp end) into the pilot hole, and trigger the drill slowly. Do not let it "run away," or the hanger bolt may be screwed in too deeply. It might even leave the pilot hole altogether, resulting in a slanted (and hence, useless) pivot. Allow the hanger bolt to enter the slat end up to the bottom of the metal "band" (if it has one) which separates the screw-end from the bolt-end. If it has no band, mark a length about a third of the way from the tip of the screw-end. Be



A drawing of the PRASP and its dimensions.

careful in storing the slats once the hanger bolts have been inserted. If they stand on end too long on a hard surface, the bolts are liable to bend.

FINISHING

Once the wood has been sanded to your satisfaction, you have three alternatives: you may use a stain varnish, which requires no top-coat, a stain and top-coat combination (probably the best from an aesthetic viewpoint) or you may simply apply clear top-coat. Regardless of the finish chosen, I recommend that the PRASP be finished in *some* way (even paint). The wood will last longer, and the unit will appear more professional. Since every finishing product is different, follow the directions on the can to ensure proper results.

As a rule, stain is applied with a piece of lint-free cloth—old undershirt or sheet material is good. Double up the cloth and dip it into the can just enough to get a thin layer of stain on the cloth. It may be a good idea to wear disposable plastic gloves, since stain is notoriously hard to wipe off. Wipe the stain *with* the grain of the wood. Work on a small area at a time. Try not to get the stain too thick in one area. Once the stain has dried, one or more coats may be applied, depending on the darkness of the shade desired.

Stain varnish and clear top-coats are usually brushed on. Again, apply with grain. Try to avoid brush marks, and blend areas together carefully.

Finishing needs to be done in a dry, well-ventilated place. If slats have carpeting attached to them while finishing, be careful not to spill or brush any finish on it (though a little varnish or top-coat carefully applied to the edge of the carpeting helps keep it from fraying.) Before doing anything with a finished piece, make sure it is totally dry. Allow at least eight hours between coats, though it is best to leave the pieces to dry overnight.

ASSEMBLY AND MOUNTING

No getting around it: putting a PRASP together is a two-person job, as is mounting it.

The easiest way to assemble all the pieces of the PRASP is sideways—that is, place the horizontal braces parallel to each other. Insert the slats one at a time into the holes of the horizontal braces *after* putting a washer, lightly lubricated with household or sewing machine oil, on each hanger bolt. You don't need to force the bolt through the pivot hole; gentle pressure is all that is required.

Once the bolt is through the hole, put on the second lightly lubricated washer and screw on (not tightly yet) the hex or wing nut. Once all slats are in place, attach the corner brackets to the horizontal braces (it helps to drill small pilot holes before driving the screws).

Next, slide vertical braces into their places between the horizontal braces. Spread some glue in the joint, and insert the end of the vertical joint; while one person holds the two braces—or while they are clamped with a corner clamp—mark and secure the other end of the corner bracket.

Repeat this procedure until all corners are bracketed on that side, flip the whole apparatus *carefully*, and repeat the procedure on all four corners. If the joints were glued, allow the unit to dry for several hours before mounting it.

Carry the PRASP over to the wall and line up the bracket holes with their respective pilot holes. While one person (or two) holds the PRASP, the other screws the bottom mounting brackets into place. Repeat the mount bracket procedure on the top. Tighten up all nuts, and test the ease of swing in the slats.

So there it is, maybe not gorgeous (if you are like me and build things only out of necessity), and then, maybe strikingly beautiful. But it's cheap, and it works. And that's what counts. ■

DIRECTORY OF STUDIO CONSTRUCTION SERVICES

Abandon/Sun, Inc.
P.O. Box 6520
San Antonio, Texas 78209
(512) 824-8781

Audio Architects, Inc.
112 Space Park Drive
Nashville, Tennessee 37211
(615) 331-3247

Audicon Marketing & Design
1200 Beechwood Avenue
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AES—Los Angeles

The Los Angeles Hilton was the scene of the 66th Audio Engineering Society Convention this past May. The convention was buzzing as usual, but this time the "gadgets" ***didn't*** *steal the show.*

At AES conventions, "excitement" is a word more often associated with the unveiling of some all-new "super-gizmo," and not with the reading of a learned technical paper. The paper may have long-term repercussions on the industry—the gizmo may be un-remembered by the next convention. Still, the paper is mostly greeted with polite applause, while the gizmo gets all the "bells and whistles."

Not so at the recent Los Angeles convention (May 6-9), where one paper stirred up the audience, and kept them that way for the duration of the show. Some people are *still* talking about it, and surely, others are hard at work in the lab, trying to prove (or disprove) the conclusions.

DIGITALIZED STRESS

In tabloid style, the paper's title stated the author's conclusion: "Human Stress Provoked by Digitalized Recordings." In the published program, convention-goers read that, "Results show that digitalized recordings cause a profound loss of muscle tone throughout the body by a factor of 300%.... The implications of these findings are quite alarming as they indicate that, should the present enthusiasm for digitalized recording continue, all of our present-day musical heritage will be recorded in a form that will be stressful to all listeners, both today and in the future."

The paper was presented by Doctor John Diamond, founder and director of the Institute of Behavioral Kinesiology. Dr. Diamond is a graduate of the Sydney (Australia) University Medical School and the author of "B.K.: Behavioral Kinesiology." For those to whom kinesiology is not yet a household word, its the "science dealing with the interrelationship of the physiological processes and anatomy of the human body with respect to movement."

A PRACTICAL DEMONSTRATION

Dr. Diamond's presentation certainly provoked its own share of Human Stress: many in the audience became quite distressed indeed by the proceedings, and by the manner in which they were conducted. Briefly, Dr. Diamond tried to make his point by testing the muscle power of several volunteers, while they (and the audience) listened to excerpts from analog and digital recordings.

The subject would extend his arm horizontally. While the music played, the volunteer would try to resist the doctor's attempt to force his arm downwards. While listening to analog recordings, the subject had no difficulty resisting Dr. Diamond's pressure: while listening to digital recordings, the doctor had no difficulty whatever, as he pushed each extended arm down.

Is this a put-on? Perhaps, but chances are it's unlikely, although the doctor's demonstration technique would have gotten him expelled from any high school lab class.

Surely, by now no one needs to be told that the body reacts—often unfavorably—to external stress-producing influences. And we all know that the mind has been alternately agitated, exhilerated, soothed, annoyed, stimulated, or put to sleep, by music. In fact, a MUZAK report states that, "Experiments performed by the Human Engineering Laboratory of the US Army...found that programmed functional music improves human vigilance, mental alertness, (and) working efficiency." Indeed, the entire background music industry owes much of its success to the influence of music on mind and body. (Better yet, erase the word "background." If *all* music didn't influence us, many of us would be unemployed.)

Dr. Diamond did caution the audience that it would be impossible to conduct a rigorous scientific inquiry in front of hundreds of curious spectators. No doubt that's quite true, but with no effort at all, he might have upgraded his presentation, and by so doing, put on a better show. For instance, a volunteer could have chosen analog and digital recordings at random, while another volunteer did the arm-pushing. That way, neither the subject nor the arm-pusher would have known what to expect in advance.

However, despite much clamor from the audience, Dr. Diamond chose to be his own D.J., M.C., *and* arm-pusher. Nevertheless, he shouldn't be tossed out on his ear, for, if he was totally "out-to-lunch" (as we say in scientific circles), surely he would not have experienced such a consistency of results. At the end of the show, the score was digital: 100%, analog, 0%, or vice-versa, depending on how you look at it.

John Woram is the editor of db Magazine, principal of Woram Audio Associates, and author of the "Recording Studio Handbook."

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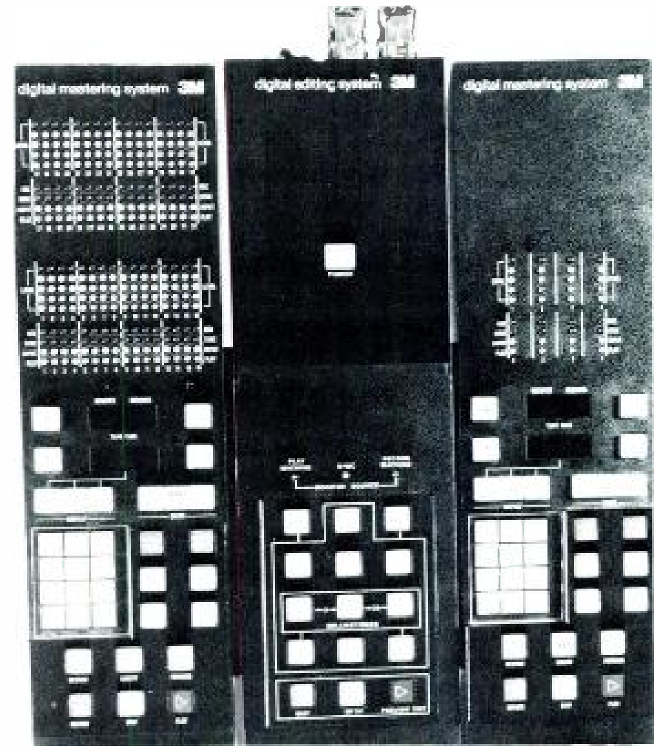
The Technic's digital editor.

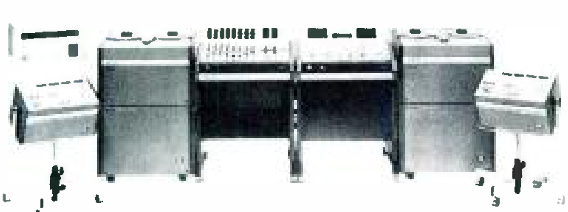


The control module of 3M's Digital Editing System offers special function buttons for determining exact editing points.

Perhaps as fascinating as the show, was the audience reaction. Some betrayed hostility out-of-proportion to the event; others saluted the doctor for revealing some profound new truth. Actually, he deserved neither the hostility nor the praise—just a word of thanks for entertaining us awhile, and reminding us that digital audio—just like analog audio, or no audio at all—will have its effect on all of us.

And, if digital audio really does relax those muscles, it may be the biggest success story since valium. Better yet, you don't need a prescription (so far). Unfortunately, Dr. Diamond did not clarify how a reduction in muscle tone could be equated with stress. It would seem, to this observer at least, the muscle power might *increase* under stress, rather than decrease. Therefore, perhaps digital audio actually reduces stress(?). Presumably, we may hope to see (and hear) of some more-conclusive tests in the not-too-distant future.

Technic's complete Digital Audio Recording System.



DIGITAL HARDWARE

Meanwhile, those with sufficient muscle-power (financial, if not physiological), had lots of digital hardware contending for their attention.

Technics demonstrated a complete digital audio recording system, consisting of four-channel tape recorders, a digital editor, a digital mixer, and a digital delay system. Not yet for sale, the purpose of the demonstration was to expose the system concept to the AES membership for reaction and comment. No doubt we may expect to see a modified system—influenced by potential-user comments—at a future convention.

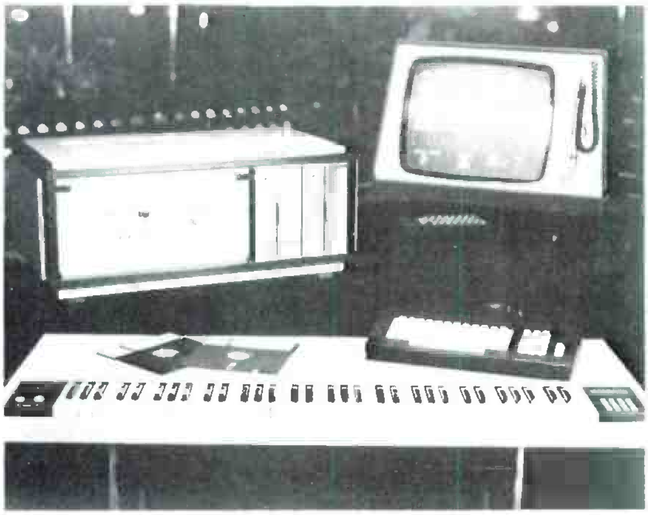
Within the complete system, the D/A conversion takes place at the front end of the mixer, not at the tape recorder. Although the mixer has analog outputs for monitoring purposes, all signal-processing is done while the signal is in the digital domain. At present, processing is limited to digital delay, although other signal-processing devices will surely be added to the system in the future.

The tape recorder (¼-inch, 15 ips, open-reel) uses 16 data tracks to record four audio channels. In addition, there are four analog channels available for recording cue tracks, time code, etc.

The digital mixer is an 8 in/2 out device, with four auxiliary outputs available to drive external signal processing devices, such as the digital delay line.

As with the Sony and JVC editing systems, the Technics editor stores several seconds of program in memory, in the vicinity of the proposed edit point. The user may then "rock" the stored edit area back and forth, to select the precise edit point.

3M displayed a production version of its digital editing system, plus a prototype cross-fade option. The cross-fade option is still under development, and will be field-tested in studios which already have digital editing systems. The cross-



The Fairlight CMI system.

fade feature, which is also found on other digital editing systems, is somewhat analogous to the diagonal splicing block in analog. Either technique allows for a smoother transition across the edit point.

Soundstream continues to offer its digital recording, editing and mastering services, rather than the sale of digital hardware. While recording and mastering services are available on-location, all editing work is done at Soundstream headquarters in Salt Lake City. Soundstream charges $1,500 per day for recording, $1,000 a day for disc mastering, and $150-per-hour for editing.

TOWARDS STANDARDIZATION?

At the convention, Sony and Studer announced a joint agreement to support a common "new format" in stationery-head digital recording. A news release states the format "...includes newly-developed and highly efficient codings for error protection and high-density recording. Sony and Studer are convinced that this new format not only provides full tape interchangeability, but also represents a major step towards the fully-digitalized audio systems of the future, with free communication between digital equipment."

Surprisingly, details of the new format were not made public at the convention, but "... will be submitted as a proposal to the appropriate industry committees in the near future."

DIGITAL MUSIC

Remember the first generation of electronic music synthesizers, with elaborate controls for setting attack, sustain, release, decay, etc? The Fairlight CMI (computer musical instrument) does away with all that drudgery by letting the musician draw a waveform on the face of a CRT monitor, using a light-pen. For those who aren't handy with a pen, a microphone is provided. Simply sing, whistle, hum, or whatever, and the computer will analyze the sound, which may then be reproduced at any pitch, using the Fairlight keyboard.

And if you're not into light-pens, microphones or piano-style keyboards, an alpha-numeric keyboard lets you use Fairlight's MCL (music composition language) software package for computer composition.

By contrast, the Crumar GDS (general development system) avoids the use of light-pens and typewriter keyboards, which the company feels are foreign to the performing musician. Instead, a small group of faders, potentiometers, joystick and foot pedals are used in conjunction with a software program for

The Bosendorfer Model 290 Imperial Grand Piano, now marketed by Kimball Piano & Organ Company.

The Shure Model A27M stereo microphone adapter permits two microphones to be mounted on a single microphone stand in a variety of selectable positions.

creating, and storing, sounds on floppy disk. A CRT screen displays control settings.

Up to 16 notes may be sounded simultaneously. Within the system, an eight-track digital sequencer may be used to record and playback keyboard movements and voicings. Sequences in memory may be played back while new ones are being created, in an almost-endless sequence of variations. Crumar plans to make available a software library of synthesized sounds for GDS users.

MORE DIGITAL MUSIC (the ten-finger type)

The legendary Bosendorfer piano is now a division of Kimball Pianos, and both were demonstrated at the convention. In Chicago, Universal Recording has recently acquired a Bosendorfer model 290 Imperial Grand piano. The model 290 is a diminutive 9'6" long (!), with 97 (!!) keys. And, its only 50+ kilo-bucks. (!!!)

With spectacular-sounding Bosendorfers finding their way into at least a few recording studios, could it be that at least a few studios will begin experimenting with spectacular-sounding mic placements? The piano sound on many of today's (non-classical) recordings can best be described as "unfortunate." Often, its not the poor piano's fault: its just been muffled, baffled and blanketed beyond recognition, with microphones tucked into the oddest places. Hopefully, the big-Bs will get better treatment, and that brings us to...

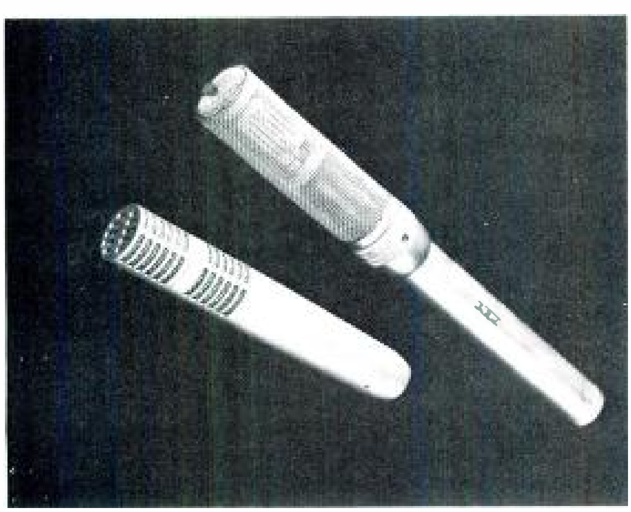
MICROPHONES AND ACCESSORIES

There seems to be a re-awakening of interest in stereo microphones and techniques. For example, Sony showed their new ECM-989, a dual-capsule M-S (middle-side) stereo microphone. The M capsule is a forward-oriented cardioid pattern, while the S capsule is a side-oriented bi-directional pattern. Varying the relative gain of the S capsule will change the directivity from mono to a 120-degree stereo pickup. The list price of the ECM-989 is $435.

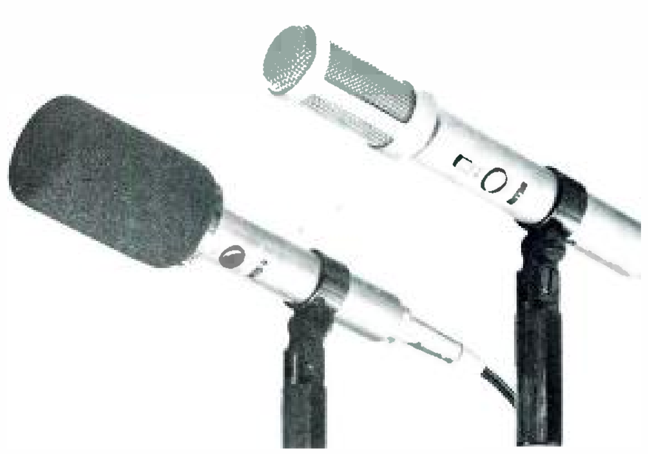
Sweden's PML Laboratories introduced two types of stereo microphones. The MXSY-8 contains two capsules, whose patterns may be varied by remote control, for M-S or X-Y operation. The XY-82 contains two cardioid capsules. If one of the capsules is rotated through 180 degrees, the combined pattern may be varied continuously from omni-to-cardioid-to-figure-8, simply by changing the console fader setting, and the relative polarity of the rotated capsule.

For a do-it-yourself M-S pickup, Audio Engineering Associates' MS 38 M S matrix decoder uses active solid-state electronics, instead of the traditional transformer. The MS 38 will work with any cardioid and figure-8 microphones. Line-

The XY-82 and MXSY-8 stereo microphones from PML.



Sony's ECM-969 and ECM-989 stereo microphones.



level inputs and outputs allow it to be conveniently patched into the console anywhere after the microphone preamplifier.

And, for holding those two microphones in place for a stereo pickup, don't overlook Shure's A27M stereo microphone adapter, which neatly solves almost all microphone alignment problems.

And, in the "well,-it's-about-time!" department, Matthews Studio Equipment, Inc. showed its complete line of long-reach (vertical) microphone stands, for hoisting any mic (stereo or otherwise) up to 22 feet. If that's not good enough, a short (6 feet) boom arm can be added. And if *that's* not enough, try a sky hook! An interesting feature of the Matthews stands is a spring-loaded guard to protect the mic-mounting threads during transit and set-up.

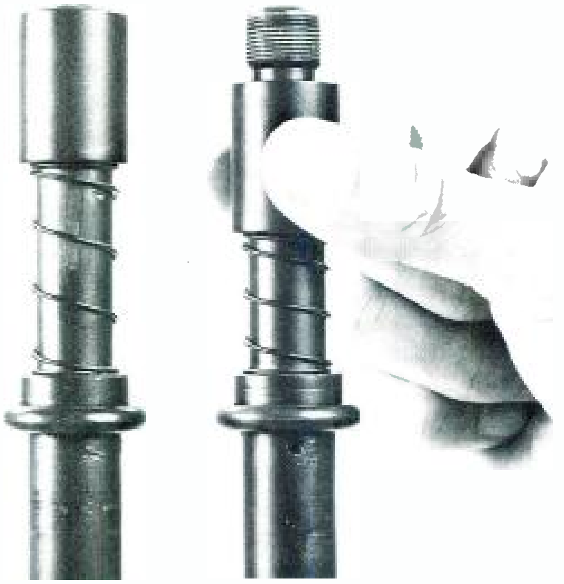
Matthews' Long-Reach microphone stands.

AMPS AND SPEAKERS

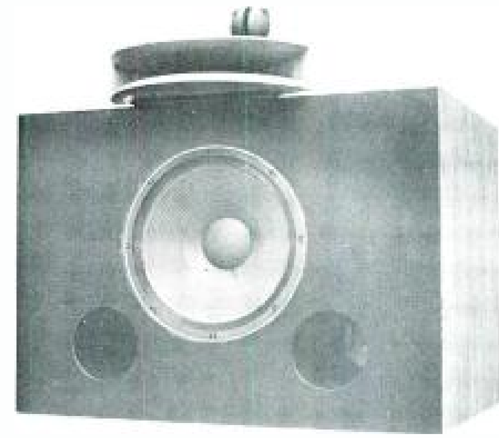
UREI enters the amplifier market with the introduction of its model 6500 dual power amplifier, which features UREI's Conductor Compensation circuit. The 6500 specs note that, "Speaker wire has significant resistance and even when fed from a 'perfect' amplifier with a '0' output impedance, degrades the system's damping factor. In addition, speaker wires have been shown to produce high frequency transient distortion due to reactive effects and non-constant group delay. (For more on damping factors, see the Sync Track column in the July 1978 db—Ed.)

The model 6500 takes the performance of the speaker wire into account in the design of the amplifier's main feedback loop. A rear-panel BNC input senses the signal at the speaker's terminals and the necessary compensation is applied. As UREI notes, the design was, "Simple in concept, but complex in execution." The 6500 delivers 275 watts into 8 ohms.

BGW introduced its largest-ever amplifier, the model 1250, which delivers 400 watts into 8 ohms. Also amplified is the



The spring-loaded Thread Guard also from Matthews'.



The Fostex LS/3 monitor system.

company's overall product line: BGW now has exclusive distribution rights for all Tannoy professional products.

Tannoy's M2000 Buckingham Monitor uses a "single Point Sound Source," that is, a dual-concentrix driver from frequencies above 350 Hz. The low end is handled by two 12-inch drivers. The system is designed for bi-amping, in conjunction with Tannoy's XO 5000 dividing network. The XO 5000 contains a three-position (20, 31.5, 63 Hz) sub-sonic filter and two single-point parametric equalizers. There is also a time control (10-400 microseconds) which allows for compensation between two drivers whose acoustic centers are up to 140 mm (5.5 inches) apart.

From Canada, Fostex showed its LS series of monitors, all of which employ the same high-frequency array. The single-element radial horn in the high-frequency section is milled from high-density teakwood. Fostex feels the LS series reduces the need for bi- and tri-amplification, since its monitors deliver two-to-four times more output (i.e., 3-to-6 dB) than older standards. Highest sensitivity comes from the LS/3 (100 dB SPL at 2.83V). Low frequency response (±3 dB) extends to: 45 Hz (LS/2), 30 Hz (LS/3), 19 Hz (LS/4).

The UREI Model 6500 power amplifier.

SIGNAL PROCESSING

Spectra Sound—a new subsidiary of Spectra Sonics—introduced three new products: the model 4000 Flanger, the 4010 Phase Shifter, and the 4020 Delay Line.

The Flanger also produces doppler, vibrato, chorus, double tracking and other effects, and may be externally controlled by foot pedals, joysticks, synthesizers, etc.

The Phase Shifter may also be externally controlled, and offers phase shifting, peak resonant phasing, and rotating speaker effects.

The Audio Delay Line will produce delays up to 120 ms. In addition, slap-back, echo, chorus and doppler effects are possible. Spectra Sound price tags are: $695 (4000), $595 (4010), and $795 (4020).

Micmix Audio Products continues to expand its line of Master Room reverberation chambers. The new top-of-the-line XL-500 system comprises three separate units: a rack-mount main controller, a smaller remote control system, and the chamber itself, which measures 22 x 22 x 62¼-inches. The XL-500 is switchable between Plate, Room and Hall modes, to provide three distinctive reverb formats. Decay times are 1-4 seconds (Plate), 2-6 seconds (Room), and 3-7 seconds (Hall). Four-band equalization is also included. Master Room systems start at $950, and the XL-500 is $4,000.

Nashville's Audicon Marketing Group showed the PARAM computer-assisted parametric equalization system—an impressive six-band parametric equalizer system capable of handling up to 128 separate channels of audio.

Instead of requiring large arrays of potentiometers, a single joystick is used to "draw" a frequency response curve on a CRT display. Next, the computer optimizes the curve, programs the selected equalizer, and displays the resultant curve. Once a satisfactory equalization has been established, the computer program may be stored on tape or floppy disk. 32 user-defined "standard" equalization settings may be stored in, and recalled from, the system memory. The computer, video display, housing for 16 channels of equalization, and power supply, costs $10,000. Each equalizer card is $1,000.

Gotham Audio's TTM Frame is a rack-mounted modular card-holding system designed to accommodate telcom, Dolby or dbx noise reduction cards. The TTM holds up to 24 carrier boards, each of which in turn holds its own noise reduction card. 15-turn potentiometers on each carrier board are used to optimize record, playback and line levels. The power supply is a separate unit, and the entire TTM system occupies 12¼-inches of rack space. ■

JOHN BORWICK

If We Can Hear It, We Can Measure It

The battle between the 'Golden Ear' faction of audio persona and the 'objectivists' rages on. As is the case in most disagreements, there are two sides, so before you line up for your "cause"—read on.

IN THE WORLD OF HI-FI, everyone is taking sides in the "Great Debate." Lined up on one side is the 'golden ear' brigade. Even when you give the brigade a frequency response dead flat from DC to infinity, and zero harmonic distortion, they can still hear "loss of information"; they have handed out blame to the amplifier (for having negative feedback that acts too slowly to cope with sharp transients), to the amplifier/loudspeaker interface (for passing signals in two ways at once), to the inch-long wires in phonograph headshells (for having too much resistance, capacitance and inductance), to all performers, capacitors and switches (for being transformers, capacitors and switches.)

THE GREAT DEBATE

Staunchly facing up to the 'golden ear' brigade are the objectivists, grimly hanging on to their sanity and muttering, "if we can hear it, we can measure it." They acknowledge that the human hearing mechanism is more complex than present methods of analysis can completely explain, but they refuse to go chasing "hi-fi" myths. They maintain that *significant* degradations in the quality of reproduced speech and music will always be traceable to known, and objectively quantifiable, causes.

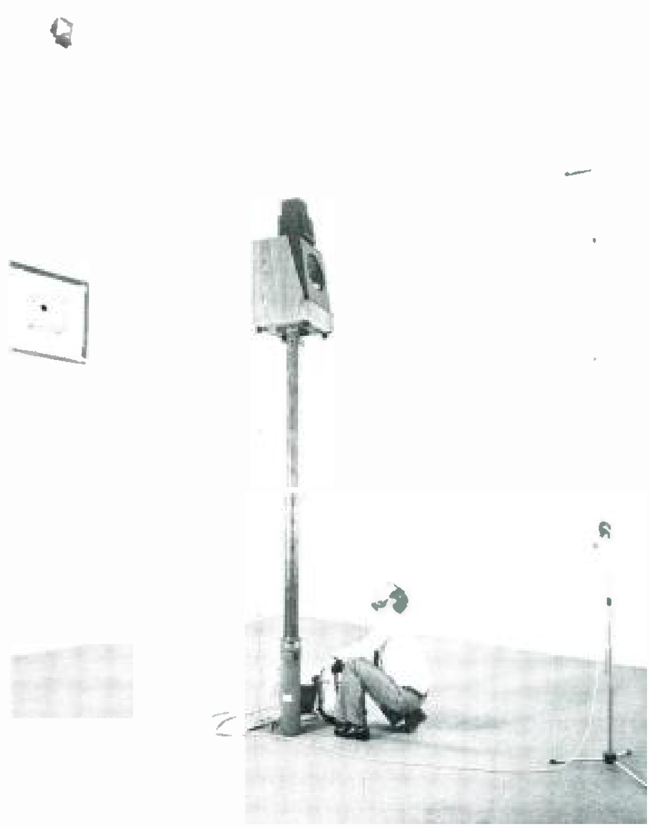
These arguments featured strongly in the papers program at the 65th AES Convention held in London in February, with both points of view represented. For instance, the very first paper was entitled "The Great Debate: Subjective Evaluation" (AES Preprint 1563). In it, the Canadian physicists Stanley Lipshitz and John Vanderkooy gave an amusing account of tests using a specially constructed A/B switch-box for 'double-blind' testing, in which two signals to be compared are randomized, and neither the listeners under test nor the test organizer knows which signal is being selected. (Compare this with the procedure described in our AES Convention Report in this issue—Ed.) They concluded that, "provided linear differences are properly removed, modern components with non-linearities which are adequately low by normal measurements, all sound alike." Taking the opposite view, the Finnish expert, Matti Otala, now working in the USA, presented two papers showing that forms of distortion not revealed in conventional measurements could be detected at even very low levels by trained listeners. (The papers are: "Feedback-generated Phase Modulation in Audio Amplifiers" (Preprint 1576), "Power Amplifier Design Parameters and Intermodulation Distortion in the Amplifier/Loudspeaker Interface" (Preprint 1608), and also the earlier 1978 "Intermodulation Distortion in the Amplifier/Loudspeaker Interface" (Preprint 1336).—Ed.)

Up till now, it has been possible for professional recording engineers to shrug off this Great Debate. It has mainly been hot news (and hot business on the domestic hi-fi battleground and, except for occasional music producers who have insisted that they dislike the "sound" of certain consoles etc., etc. the studio scene has been able to carry on as usual.

John Borwick is a freelance writer and author of "Sound Recording Practice."

Figure 1. The Laboratory listening room at KEF.

Figure 2. Acoustics lab test room.



But the question of what we can and cannot hear (or tolerate) becomes of burning interest when new international standards have to be agreed—as at the present stage of development of digital recording. Certain 'golden ear' factions are insisting that we shoot for very-perfectionist figures for both frequency range and dynamic range—say DC to 20 kHz, and 96 dB respectively. Yet, this kind of bandwidth and peak signal-to-noise ratio puts up the sampling frequency, the bit numbers—and, more important, the cost—enormously. It could even delay the wider acceptance of digital recording, and this would be a pity, considering digital's many inherent advantages.

CRITERIA NEEDED

To arrive at a reasonable standards specification, we need to know with a good degree of certainty (a) what is the narrowest bandwidth tolerated by most average listeners, and (b) what degree of peak clipping is acceptable?

Perhaps with this in mind, the AES British Section, who organized the 65th Convention, set a room aside for special "Subjective Test" sessions under control conditions. The tests were sponsored by KEF Electronics, the loudspeaker manufacturers, and master-minded by Laurie Fincham, the Convention Chairman, and also KEF's technical director. The tests were aimed directly at checking the audibility of the upper cut-off frequency of filters, and soft and hard clipping in amplifiers.

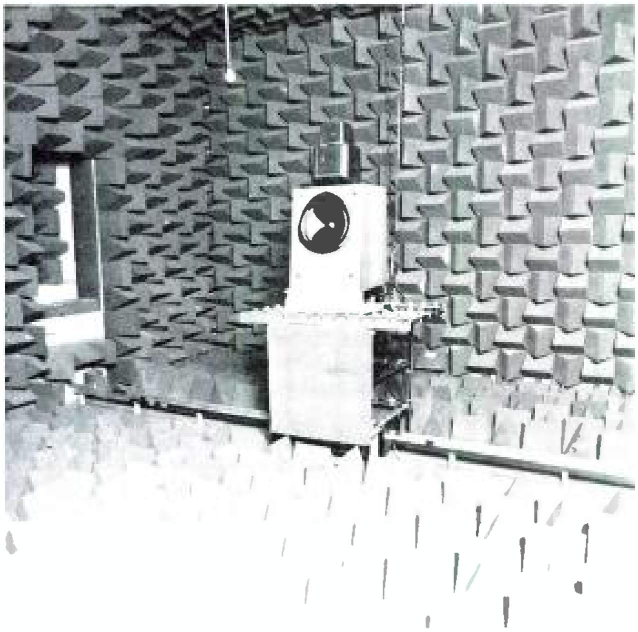


Figure 3. The production test chamber.

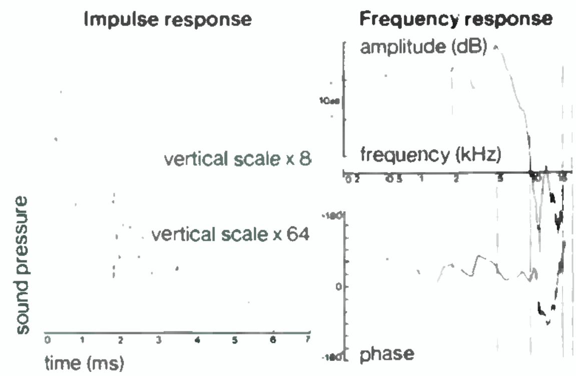


Figure 4. A chart of impulse response contrasting time and sound pressure; and a frequency response graph which depicts amplitude, frequency, and phase.

The signal replay chain was single channel, with a choice of program sources and a single full-range monitoring loudspeaker. The signal could be switched at random through a distorting or a non-distorting channel by means of a programmable scanner and multiplexer (PSM), which was under computer control. Speaker input and output levels were closely monitored from test to test, and each program item was played twice and indicated as 'A' or 'B' on a computer controlled light panel. A 'Vote Now' display followed each A B playing and each member of the listening panel (up to a maximum of 10 participants) could then press individual hand-held YES NO buttons to show whether he could hear any difference. In a number of pairings, the undistorted (control) signal was used as both A and B.

For the filter tests, the test signal was a differentiated 1 kHz square wave, whose spectrum rolled off above 50 kHz. The filters were 9th-order elliptical, having cut-off frequencies of 10, 16 and 20 kHz. For the clipping tests, piano examples on a Sony PCM 1600 digital recorder were used, through special circuits designed to give 'hard' and 'soft' clipping characteristics while maintaining constant overall gain.

The results, analyzed got about 60 listeners, and were briefly outlined by Laurie Fincham at a technical forum on the last afternoon of the convention. They appeared to support the objectivist cause: while a majority of listeners could detect the 10 kHz filter, very few (less than 5 percent) could be sure at 16 kHz, and hardly anyone at 20 kHz. Similarly, the soft clipping scores were not much better than would be obtained by chance, though the hard clipping could be heard by most trained listeners.

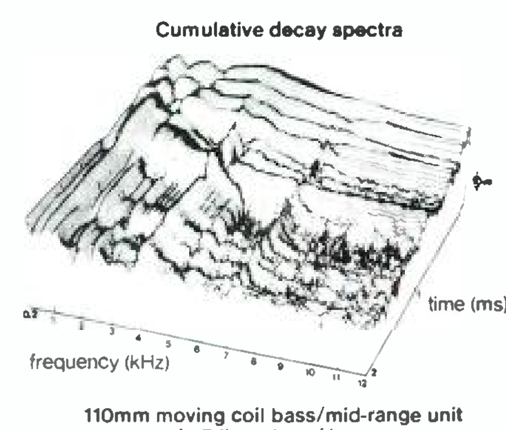
A VISIT TO THE KEF LABORATORIES

Besides setting up this test at the AES Convention, Laurie Fincham has a deep interest in proper subjective testing, and is on two IEC (International Electrotechnical Commission) Working Groups, devoted to Loudspeakers and Listening Tests. So, I went down to KEF Laboratories a few days after the convention to see how much importance was placed on listening—as opposed to the complex computer-aided measurement procedures for which KEF are already famous.

I was not disappointed. The laboratory listening room (FIGURE 1) must be as good as you will find anywhere. It conforms in all details of dimensions, reverberation time frequency curve and positional uniformity, with the latest IEC proposals, and easily exceeds them in terms of insulations against outside noise. The building is isolated by floating it on a 260-ton concrete raft, surrounded by a deep trench filled with large pebbles. More important, it makes an ideal venue for analytical listening to recorded music, as opposed to quality assessment of equipment. (I was able to confirm this on a separate visit, when I convened a whole day's listening at KEF for my panel of music and technical critics to produce our quarterly "Sounds in Retrospect" review of disc quality for *Gramophone*.)

KEF mainly used the room for subjective testing of design prototypes, sample speakers from the production line, and 100 percent system-testing of their "prestige" models like the Model 105 Series II. One clever test uses male speech recorded in the

Figure 5. A cross-sectional depiction of cumulative decay spectra when testing a 110 mm moving coil bass/mid-range unit in a 7 litre closed box.



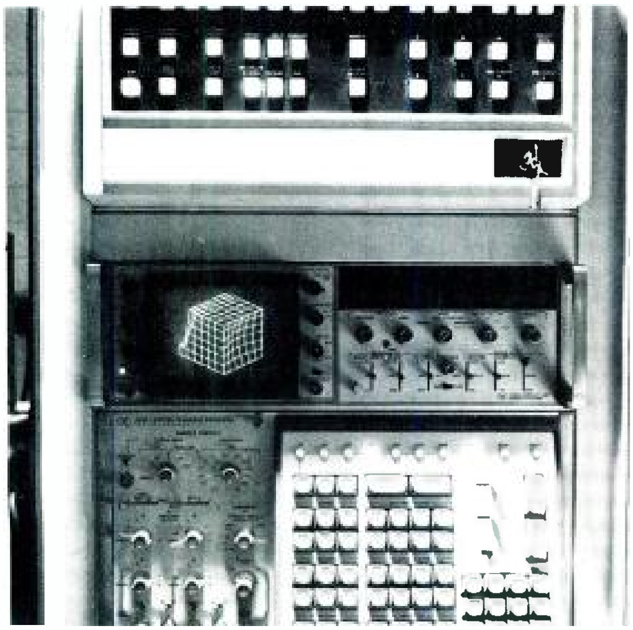
anechoic test chamber by one of the engineers, whose voice is well-known to his colleagues. When this engineer then takes up various positions behind the front-lit curtain and interpolates the speech phrases 'live' between those reproduced through the speakers under test, critical comparisons can be made. This assists in checking the matching of left and right speakers in a stereo pair, as does careful listening to switched pink-noise signals, and ties in with what is perhaps the unique feature of the KEF objective test procedures.

Every drive unit and crossover network is individually computer-tested before being matched with its stereo partner—and then the assembled enclosures are similarly tested. The procedure for a complete system, for example, is to place it near the center of a room with good insulation against external noise and feed it with a stream of short electrical impulses. The room need not be anechoic, but it should be large enough for the vibrations of the speaker to die away before the arrival of the first boundary reflection. FIGURE 2 shows the test room used in system research and development. It has a reverberation time of about four seconds. To speed up the measurements of the prestige systems on the production line, a further test room is used (FIGURE 3) with a reverberation time of 0.2 seconds.

The acoustic response of the speaker is picked up by a microphone, digitized and stored in a digital processor. It takes the form of a strong initial pulse, and stored in a digital processor. It takes the form of a strong initial pulse, followed by a number of diminishing peaks. In practice, a number of repetitions—up to 500—is averaged to enhance the signal-to-noise ratio and permit close scrutiny of the low level tail of the impulse.

The impulse response may be viewed directly, or it can be used to derive the more usual amplitude/frequency and phase/frequency responses by a mathematical process known as the Fast Fourier Transform (FIGURE 4). The method avoids the need for anechoic testing, which is in any case time-consuming and suspect at low frequencies, but has important side advantages. For example, further mathematical programming can produce the now-familiar three-dimensional cumulative spectra of a speaker's response in terms of amplitude, frequency and time during build up and decay (FIGURE 5). This can reveal defects not identifiable by other means, and greatly assists the proper diagnosis of cause and effect. In another application, I was shown how modal analysis actually displays a graphical representation of the cabinet walls (FIGURE 6). The display is a computer-controlled replay of data stored from numerous contact-microphone recordings. The surfaces are seen to flex under the action of the drive signal, and the display pinpoints the best positions for internal bracing and other structural features to minimize resonances.

Figure 6. Model analysis of cabinet motion.



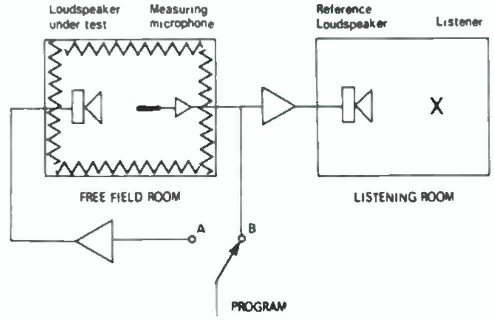


Figure 7A. A combination of objective and subjective techniques which were developed by KEF for digital simulation of an 'imaginary' loudspeaker systems.

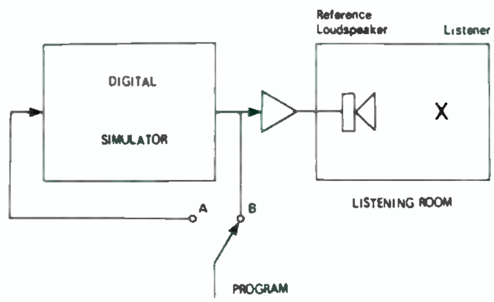


Figure 7B. The source material can be switched directly to the reference loudspeaker, with no audible difference from the subjectively "perfect" loudspeaker.

HARNESSING THE COMPUTER TO LISTENING TESTS

A coming-together of objective and subjective techniques has also been pioneered at KEF in the digital simulation of 'imaginary' loudspeaker systems. Look first at the test set-up of FIGURE 7A. The speaker under test is placed in the room used for objective measurements and its output relayed, via the measuring microphone and a reference loudspeaker, to a listener in a typical listening room. The source material can be switched directly to the reference loudspeaker (FIGURE 7B) or via the test loudspeaker/microphone chain (A). If the signal levels are kept the same, there will be no audible differences for a subjectively 'perfect' loudspeaker. When differences can be heard, the test loudspeaker and microphone chain is replaced by a digitally-realized system having the same measured impulse response. The test loudspeaker output can be simulated digitally by processing the signal material using a computer (and stored impulse response of the loudspeaker). It is then possible to compare the quality of the straight program and the simulated loudspeaker—and simulate any proposed modifications to the design without having to make any physical changes or new prototypes.

Surely these painstaking tests, relying equally on ears and the best available objective techniques (by the way, I should mention that Laurie Fincham and several of his colleagues are performing musicians as well as engineers) point the way ahead for improved audio components, and digital media in the studios and in the home. ■

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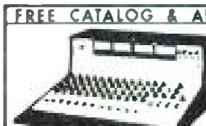

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People/Places/Happenings

• **Jack Clement Recording Studios** of Nashville has changed its name. As of May 1st of this year, Jack Clement Recording is now to be known as **Sound Emporium**. The two-studio, 24-track facility was purchased from founder, Jack Clement, by producer **Larry Butler** and financial consultant **Al C. Mifflin** in 1975. The new owners decided to retain the name and its associated goodwill until recently. Sound Emporium has enjoyed a "hitmaker" reputation for years, recording such notables as the **Amazing Rhythm Aces, Julie Andrews, Don McLean, Kenny Rogers, Joe Stampley, Waylon Jenning, Willie Nelson, and Don Williams.** During the first three months of 1980, the studio averaged 14 per cent of the singles and 21 per cent of the albums on the three major charts each week.

• **Bruno Spoerri Recordings** (Zurich, Switzerland) have just completed updating the studio with a new MCI 536 computerized console, an MCI JH-110A 2-track with AutoLocator, the new Studer A800 24-track machine also with AutoLocator, and the AMS digital delay and pitch shifter and the Lexicon 224 digital reverb. The studio has the widest variety of synthesizers in Switzerland, including the EMS Synthi 100, Prophet-5, different Lyricons, ARP & Moog synthesizers etc. Now in planning is a new large studio with board and lodging facilities. It will be built into a farmhouse which is situated in the Swiss countryside near Zurich.

• **Andre Perry's Le Studio** is bursting with energy these days. They've just installed a $300,000 computerized **Solid State Logic** console. **Nick Blagona** is nominated for the engineering award at this year's Juno (Canada's Grammy) awards. Perry is nominated for producing **Wilson Pickett, Rush, April Wine, Streetheart, Teaze, Minglewood**—all nominees at the awards ceremony this year, chose Le Studio for their latest recordings. The studio is located in Morin Heights, Quebec, Canada.

• **James B. Lansing Sound, Inc.'s Operations Division** has been expanded with the appointments of **Jerry Feingold** as director of manufacturing engineering and **Ray Blinde** as director of materials management. In his new position, Feingold will oversee the entire manufacturing engineering department. In turn, Blinde will be responsible for coordinating product development and master scheduling. Feingold received a Bachelor of Science degree in Industrial Engineering and a Master's degree in Business Administration from Fairleigh Dickinson University. Blinde holds a Bachelor's degree in Business Administration from Colorado State University.

• **Tony Hartin** has been appointed sales training manager for **Sony's Audio Division**. Hartin's responsibilities include sales training on all new and current hifi equipment, sales presentations and close contact with Sony's sales force and dealer network. Prior to joining Sony, he served as national training manager for **Technics**.

• **Lewis & Associates Public Relations** has announced the appointment of **Tarra Thomas** to the position of vice president and account executive. Prior to joining Lewis & Associates, Thomas was a partner in a public relations firm. Aside from consumer electronics, Thomas has a wide variety of experience in entertainment, home furnishings and leisure time industries.

• **Joseph A. Palmieri** has recently rejoined the **Bogen Division of Lear Siegler, Inc.**, as vice president of marketing. Palmieri served Bogen as director of sales from 1977 to 1979, leaving to become vice president of marketing for **Lebo Products** of New York. In his new position, he will be responsible for marketing planning, sales, advertising, promotion and all aspects of distributor relations. The new vice president holds a Bachelor of Arts degree from Bloomfield College.

• **Van Webster**, owner of **Hope Street Studio**, announced the opening of **Digital Sound Recording**, a new full service digital sound recording company. Services to be offered by Digital include original digital recordings in their 24-track studio, remote digital recording service worldwide, 24-track mixdowns to 2-track digital masters, editing, assembly and preparation of digital masters, and digital playback for disc mastering. Other offerings include making its complete system available on a rental basis with a digital technician for recording in outside studios. Digital Sound Recording is located at 607 North Avenue 64, Los Angeles, Ca.

• **Ackerman & McQueen Advertising** has recently added equipment to its computerized 24-track **"Golden Voice"** recording studio. Latest additions include a Neumann KM 84 and an Electro-Voice RE-20. A Teletronix LA-2A tube limiter has been added to its selection of UREI limiters and four more parametric channels have been added to its MCI 600 series console, making 14 parametric and 14 normal channels. The studio also boasts a complete set of natural wood Yamaha drums.

• **Switchcraft, Inc.**, of Chicago, has announced the appointment of **Eric Long** as director of engineering. He replaces **Paul Hoppe, Jr.**, who was promoted to vice president of marketing of the electronics manufacturing company. Prior to joining Switchcraft, Long held engineering management positions with **Allen Bradley** and **Cutler Hammer.** Long holds more than 50 patents pertaining primarily to electrical and electromechanical devices.

• **Telarc Digital Records** announces the appointment of **Michael Kellman** to the newly created position of director of marketing. Kellman comes to Telarc with a broad background in both classical record and audiophile equipment marketing and sales. Telarc Digital Records markets their records via **Audio Technica's** distributors.

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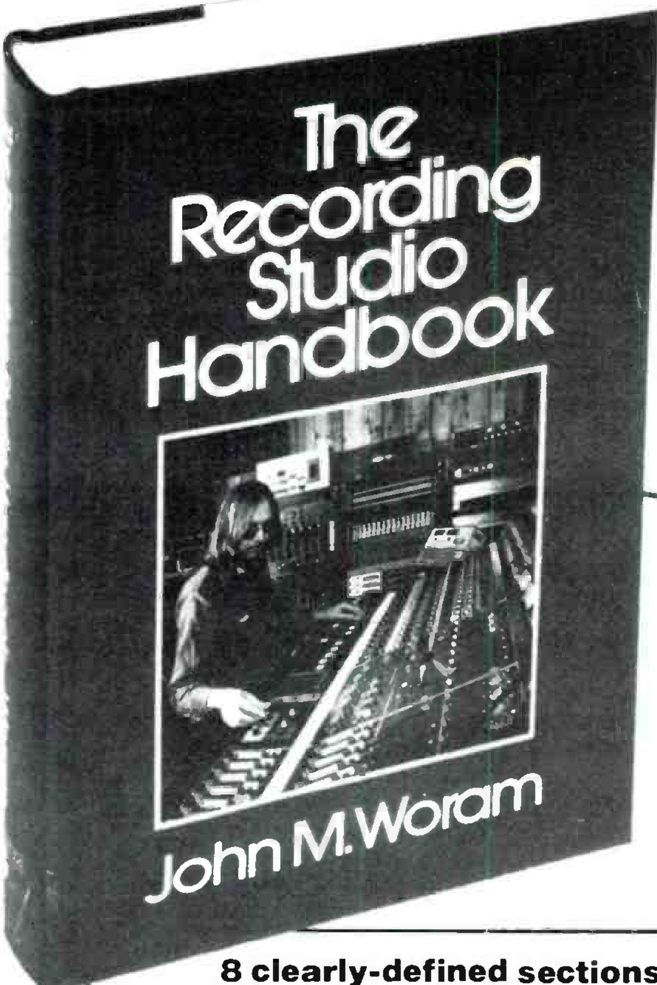

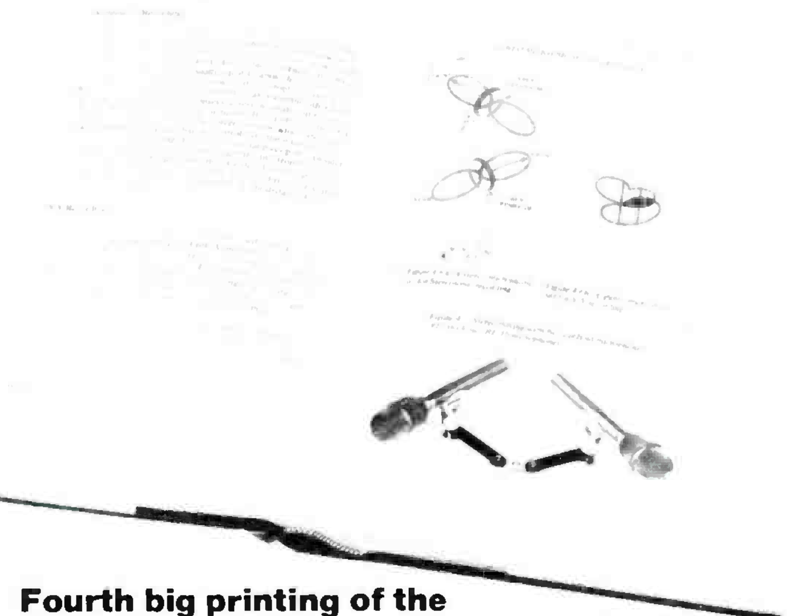

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